MultiVOIP®

Voice / Fax over IP Networks

User Guide for Voice/IP Gateways

ISDN/BRI Models: MVP410ST
MVP810ST

MultiTech® Systems

User Guide

S000385B

ISDN/BRI MultiVOIP Units (Models MVP410ST & MVP810ST)

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Chapter 1: Overview

About This Manual

This manual is about Voice-over-IP products made by Multi-Tech Systems, Inc. It describes two ISDN/BRI MultiVOIP® units, models MVP810ST and MVP410ST. The MVP810ST has four ISDN/BRI interfaces and therefore eight ISDN B-channels; the MVP410ST has two ISDN/BRI interfaces and therefore four ISDN B-channels.

These MultiVOIP units can inter-operate with other contemporary analog MultiVOIP units (MVP810, MVP410, MVP210, MVP130 & FXS/FXO MultiVOIPs like the MVP130FXS, MVPFX2-2/4/8 and MVPFXS-8/16/24), with contemporary digital T1/E1/ISDN-PRI MultiVOIP units (MVP2410 and MVP3010), and with the earlier generation of MultiVOIP products (MVP200, MVP400, MVP800, MVP120, etc.)

The table below (on next page) describes the vital characteristics of the various models described in this manual.

How to Use This Manual. ***In short, use the index and the examples.*** When our readers crack open this large manual, they generally need one of two things: information on a very specific software setting or technical parameter (about telephony or IP) *or* they need help when setting up phonebooks for their voip systems. The index gives quick access to voip settings and parameters. It's detailed. Use it. The best way to learn about phonebooks is to wade through examples like those in our chapters on T1 (North American standard) Phonebooks and E1 (Euro standard) Phonebooks. Note that we have chapters on both T1 phonebooks and E1 phonebook even though this manual is about ISDN-BRI voip units. "T1" simply refers to North American telephony standards, whereas "E1" refers to European telephony standards.

The printed Quick Start Guide shipped with your MultiVOIP (and included on the MultiVOIP product CD in electronic form) contains quick setup information for a minimal voip configuration.

Finally, this manual is meant to be comprehensive. If you notice that something important is lacking, please let us know.

Additional Resources. The MultiTech web site (www.multitech.com) offers both a list of Frequently Asked Questions (the MultiVOIP FAQ) and a collection of resolutions of issues that MultiVOIP users have encountered (these are Troubleshooting Resolutions in the searchable Knowledge Base).

The table below summarizes the features of the MultiVOIP units described in this manual.

MultiVOIP BRI Product Family			
Description / Model	**MVP810ST**	**MVP410ST**	
Function	ISDN-BRI voip	ISDN-BRI voip	
Capacity	4 ISDN lines (8 B-channels)	2 ISDN lines (4 B-channels)	
Chassis/ Mounting	19” 1U rack mount	19” 1U rack mount	
	1. “BRI” means Basic Rate Interface.		

Introduction to ISDN-BRI MultiVOIPs (MVP410ST & MVP810ST)

VOIP: The Free Ride. We proudly present Multi-Tech's MVP-410ST/810ST generation of MultiVOIP Voice-over-IP Gateways. All of these models allow voice/fax communication to be transmitted at no additional expense over your existing IP network, which has ordinarily been data only. To access this free voice and fax communication, you simply connect the MultiVOIP to your telephone equipment and your existing Internet connection. These ISDN Basic Rate Interface (ISDN-BRI) MultiVOIPs inter-operate readily with T1 or E1 MultiVOIP units (T1 and E1 MultiVOIP units can operate in ISDN Primary Rate Mode, ISDN-PRI, as well).

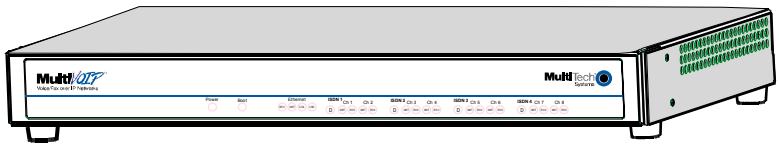


Figure 1-1: MVP-410ST/810ST Chassis

Capacity. MultiVOIP model MVP810ST accommodates four ISDN-BRI lines (eight B-channels) and model MVP410ST accommodates two ISDN-BRI lines (four B-channels). Both of these MultiVOIP units have a 10/100Mbps Ethernet interface and a command port for configuration.

Mounting. Mechanically, the MVP410ST and MVP810ST MultiVOIPs are designed for a one-high industry-standard EIA 19-inch rack enclosure. The product must be installed by qualified service personnel in a restricted-access area, in accordance with Articles 110-16, 10-17, and 110-18 of the National Electrical Code, ANSI/NFPA 70.

Phone System Transparency. These MultiVOIPs inter-operate with a telephone switch or PBX, acting as a switching device that directs voice and fax calls over an IP network. The MultiVOIPs have "phonebooks," directories that determine to who calls may be made and the sequences that must be used to complete calls through the MultiVOIP. The phonebooks allow the phone user to interact with the VOIP system just as they would with an ordinary PBX or telco switch. When the phonebooks are set, special dialing sequences are minimized or eliminated altogether. Once the call destination is determined, the phonebook settings determine whether the destination VOIP unit must strip off or add dialing digits to make the call appear at its destination to be a local call.

H. 323, SIP, & SPP. Being H.323 compatible, the BRI MultiVOIP unit can place calls to telephone equipment at remote IP network locations that also contain H.323 compatible voice-over-IP gateways. It will interface with H.323 software and H.323 gatekeeper units. H.323 specifications also bring to voip telephony many special features common to conventional telephony. H.323 features of this kind that have been implemented into the MultiVOIP include Call Hold, Call Waiting, Call Identification, Call Forwarding (from the H.450 standard), and Call Transfer (H.450.2 from H.323 Version 2). The fourth version of the H.323 standard improves system resource usage (esp. logical port or socket usage) by handling call signaling more compactly and allowing use of the low-overhead UDP protocol instead of the error-correcting TCP protocol where possible.

The MultiVOIP is also SIP-compatible. ("SIP" means Session Initiation Protocol.) However, H.450 Supplementary Services features can be used under H.323 only and not under SIP.

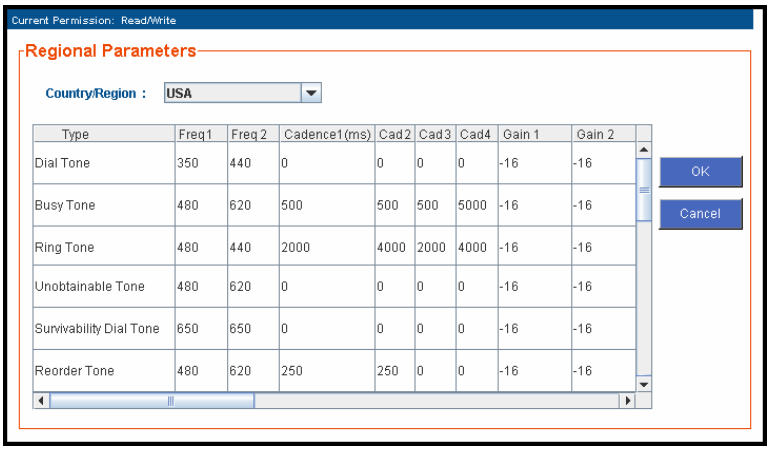
SPP (Single-Port Protocol) is a non-standard protocol developed by Multi-Tech. SPP is not compatible with the "Proprietary" protocol used in Multi-Tech's earlier generation of voip gateways. SPP offers advantages in certain situations, especially when firewalls are used and when dynamic IP address assignment is needed. However, when SPP is used, certain features of SIP and H.323 will not be available and SPP will not inter-operate with voip systems using H.323 or SIP.

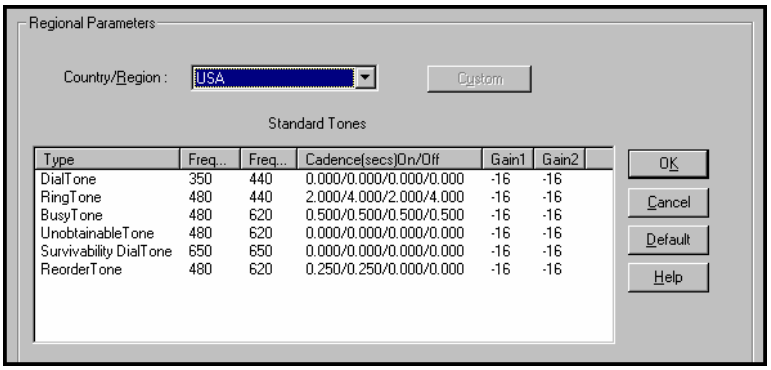
Data Compression & Quality of Service. The BRI MultiVOIP unit comes equipped with a variety of data compression capabilities, including G.723, G.729, and G.711 and features DiffServ quality-of-service (QoS) capabilities.

Gatekeepers. For voip systems built with MultiTech's analog gateway units, users can have a stand-alone gatekeeper (gatekeeper software residing in separate hardware). Gatekeepers are optional but useful within voip systems. The gatekeeper acts as the 'clearinghouse' for all calls within its zone. MultiTech's stand-alone gatekeeper software performs all of the standard gatekeepers functions (address translation, admission control, and bandwidth control) and also supports many valuable optional functions (call control signaling, call authorization, bandwidth management, and call management).

Management. Configuration and system management can be done locally with the MultiVOIP configuration software. After an IP address has been assigned locally, other configuration can be done remotely using the MultiVOIP web browser GUI. (The default IP address for the BRI MultiVOIP unit is 192.168.3.143.) Remote system management can be done via the MultiVOIP web browser GUI. The MultiVOIP configuration software pack is included on the Product CD and the web browser accesses firmware in the MultiVOIP unit itself.

While the web GUI's appearance differs slightly, its content and organization are essentially the same as that of the Windows GUI (except for logging).

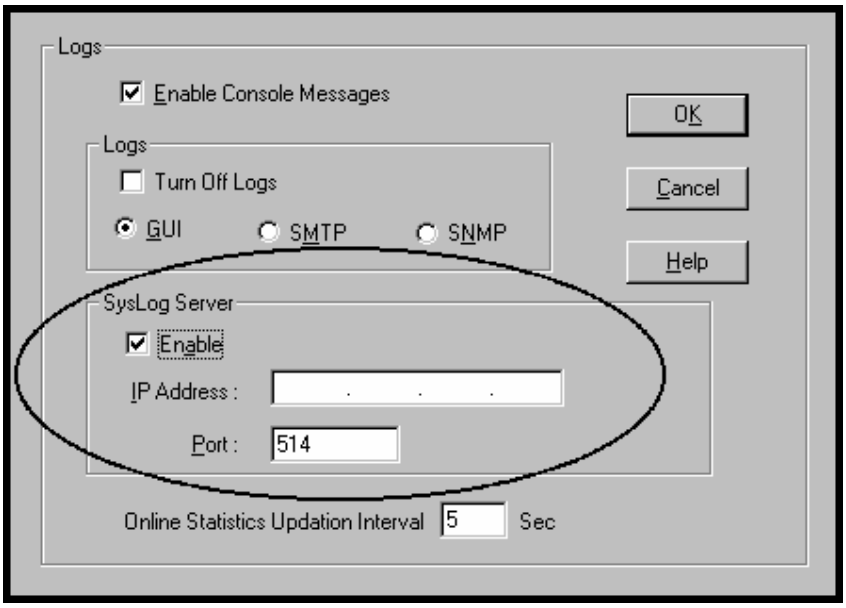

Current Permission: Read/Write
Regional Parameters
Country/Region : USA
Type Freq 1 Freq 2 Cadence1(ms) Cad2 Cad3 Cad4 Gain 1 Gain 2
Dial Tone 350 440 0 0 0 0 -16 -16
Busy Tone 480 620 500 500 500 5000 -16 -16
Ring Tone 480 440 2000 4000 2000 4000 -16 -16
Unobtainable Tone 480 620 0 0 0 0 -16 -16
Survivability Dial Tone 650 650 0 0 0 0 -16 -16
Reorder Tone 480 620 250 250 0 0 -16 -16
OK
Cancel



The primary advantage of the web GUI is remote access for control and configuration. The controller PC and the MultiVOIP unit itself must both be connected to the same IP network and their IP addresses must be known.

Once you've begun using the web browser GUI, you can go back to the MultiVOIP Windows GUI at any time. However, you must log out of the web browser GUI before using the MultiVOIP Windows GUI.

Logging of System Events. MultiTech has built SysLog Server functionality into the software of the MultiVOIP units. SysLog is a *de facto* standard for logging events in network communication systems.



The SysLog Server resides in the MultiVOIP unit itself. To implement this functionality, you will need a SysLog client program (sometimes referred to as a "daemon"). SysLog client programs, both paid and freeware, can be obtained from Kiwi Enterprises, among other firms. See www.kiwisyslog.com. SysLog client programs essentially give you a means of structuring console messages for convenience and ease of use.

MultiTech Systems does not endorse any particular SysLog client program. SysLog client programs by any qualified provider should suffice for use with MultiVOIP units. Kiwi's brief description of their SysLog program indicates the typical scope of such programs. "Kiwi Syslog Daemon is a freeware Syslog Daemon for the Windows platform. It receives, logs, displays and forwards Syslog messages from hosts such as routers, switches, Unix hosts and any other syslog enabled device. There are many customizable options available."

Supplementary Telephony Services. The H.450 standard (an addition to H.323) brings to voip telephony more of the premium features found in PSTN and PBX telephony. MultiVOIP units offer five of these H.450 features: Call Transfer, Call Hold, Call Waiting, Call Name Identification (not the same as Caller ID), and Call Forwarding. (The first four features are found in the "Supplementary Services" window; the fifth, Call Forwarding, appears in the Add/Edit Inbound phonebook screen.) Note that the first three features are closely related. All of these H.450 features are supported for H.323 operation only; they are *not* supported for SIP or SPP.

ISDN BRI MultiVOIP Front Panel LEDs

LED Types. The MultiVOIPs have two types of LEDs on their front panels:

(1) general operation LED indicators (for power, booting, and ethernet functions), and
(2) channel operation LED indicators that describe the data traffic and performance in each VOIP data channel.

Active LEDs. On the MVP810ST, there are four sets of ISDN-operation LEDs. On the MVP410ST, there are two sets of ISDN-operation LEDs. Each set contains one "D" LED and two sets of channel operation LEDs (XMT and RCV).

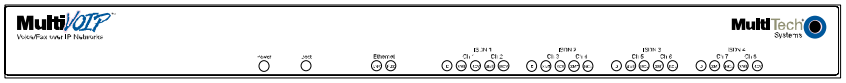


Figure 1-2. MVP-410ST/810ST LEDs

ISDN-BRI MultiVOIP LED Descriptions

MVP-410ST/810ST Front Panel LED Definitions	
LED NAME	**DESCRIPTION**
General Operation LEDs (one set on each MultiVOIP model)	
Power	Indicates presence of power.
Boot	After power up, the Boot LED will be on briefly while the MultiVOIP is booting. It lights whenever the MultiVOIP is booting or downloading a setup configuration data set.
Ethernet	**FDX.** LED indicates whether Ethernet connection is half-duplex or full-duplex (FDX) and, in half-duplex mode, indicates occurrence of data collisions. LED is on constantly for full-duplex mode; LED is off constantly for half-duplex mode. When operating in half-duplex mode, the LED will flash during data collisions. **LNK.** Link/Activity LED. This LED is lit if Ethernet connection has been made. It is off when the link is down (i.e., when no Ethernet connection exists). While link is up, this LED will flash off to indicate data activity.
D-Channel Operation LEDs (one for each ISDN line)	
D	ISDN D-channel & physical layer indicator. One "D" LED for each ISDN-BRI connection. The "D" LED is off when the BRI physical layer is de-activated.* It flashes when a connection is being established on the physical layer. It is on when the physical layer has been activated. It flickers to indicate D-channel traffic. *If the voip is running in terminal mode and its BRI line is unplugged, the D LED goes off. However, if the voip is running in network mode and its BRI line is unplugged, its LED will flash at regular interval.
B-Channel Operation LEDs (one for each B-channel)	
XMT	**Transmit**. This indicator blinks when voice packets are being transmitted onto the B-channel.
RCV	**Receive**. This indicator blinks when voice packets are being received on the B-channel.

Computer Requirements

The computer on which the MultiVOIP's Windows configuration program is installed must meet these requirements:

- must be IBM-compatible PC with MS Windows operating system;
- must have an available COM port for connection to the MultiVOIP.

However, this PC does not need to be connected to the MultiVOIP permanently. It only needs to be connected when local configuration and monitoring are done.

The MultiVOIP can also be configured and monitored using the MultiVOIP web-based configuration program. In this case, the command computer does not need to be an IBM-compatible PC nor does it need to be running a Microsoft Windows operating system. The MultiVOIP web-based configuration program can be accessed by a web browser running on any operating system (for example, Linux, MacIntosh, or IBM-compatible computers). To access the MultiVOIP by its web GUI, use the IP address 192.168.3.143, which is the MultiVOIP's default IP address. (For permanent use, you will need to change the IP address from the default value to a value that fits your voip system.)

Specifications

Parameter ……/Model	MVP410ST	MVP810ST
Operating Voltage/ Current	100-240VAC 1.2-0.6 A	100-240VAC 1.2-0.6 A
Mains Frequencies	50/60 Hz	50/60 Hz
Power Consumption	12 watts	18 watts
Mechanical Dimensions	1.75″ H x 17.4″ W x 8.5″ D 4.5cm H x 44.2 cm W x 21.6 cm D	1.75″ H x 17.4″ W x 8.5″ D 4.5cm H x 44.2 cm W x 21.6 cm D
Weight	6.61 lbs. (3.00 kg)	6.75 lbs. (3.06 kg)

Installation at a Glance

The basic steps of installing your MultiVOIP network involve unpacking the units, connecting the cables, and configuring the units using management software (MultiVOIP Configuration software) and confirming connectivity with another voip site. This process results in a fully functional Voice-Over-IP network.

Related Documentation

The MultiVOIP is shipped with a printed *Cabling Guide* that is meant simply to get the unit connected properly. The MultiVOIP User Guide (the document you are now reading) comes in electronic form and is included on your system CD. It presents in-depth information on the features and functionality of Multi-Tech's MultiVOIP Product Family. The "Quick Start Instructions" chapter of this manual describes how to get the manual up and running.

The CD media is produced using Adobe Acrobat™ for viewing and printing the user guide. To view or print your copy of a user guide, load Acrobat Reader™ on your system. The Acrobat Reader is available as a free download from Adobe's Web Site:

www.adobe.com/prodindex/acrobat/readstep.html

This MultiVOIP User Guide is also available on Multi-Tech's Web site at:

http://www.multitech.com

Viewing and printing a user guide from the Web also requires that you have the Acrobat Reader loaded on your system. To select the MultiVOIP User Guide from the Multi-Tech Systems home page, click **Documents** and then click **MultiVOIP Family** in the product list drop-down window. All documents for this MultiVOIP Product Family will be displayed. You can then choose *User Guide (MultiVOIP Product Family)* to view or download the **.pdf** file.

Entries (organized by model number) in the "knowledge base" and 'troubleshooting resolutions' sections of the MultiTech web site (found under "Support") constitute another source of help for problems encountered in the field.

Chapter 2: Quick Start Instructions

Introduction

This chapter will get your MultiVOIP up and running quickly. For more details, see the relevant specific chapters of this manual.

Safety Warnings

Lithium Battery Caution

A lithium battery on the voice/fax channel board provides backup power for the timekeeping capability. The battery has an estimated life expectancy of ten years.

When the battery starts to weaken, the date and time may be incorrect. If the battery fails, the board must be sent back to Multi-Tech Systems for battery replacement.

Warning: There is danger of explosion if the battery is incorrectly replaced.

Safety Warnings Telecom

1. Never install telephone wiring during a lightning storm.
2. This product must be disconnected from power source and telephone network interface when servicing.
3. This product is to be used with UL and cUL listed computers.
4. Never touch uninsulated telephone wires or terminals unless the telephone line has been disconnected at the network interface.
5. Use caution when installing or modifying telephone lines.
6. Avoid using a telephone (other than a cordless type) during an electrical storm. There may be a remote risk of electrical shock from lightning.
7. Do not use a telephone in the vicinity of a gas leak – not even to report a gas leak.
8. To reduce the risk of fire, use only 26 AWG or larger telecommunication line cord.
9. Never install a telephone jack in wet locations unless the jack is specifically designed for wet locations.

MultiVOIP Startup Tasks

Task	Summary
● **Collecting Phone/IP Details (*vital!*)**	The MultiVOIP must be configured to interface with your particular phone system and IP network. To do so, certain details must be known about those phone and IP systems.
● **Placement**	Decide where you'll mount the voip.
● **Command/Control Computer Setup: Specs & Settings**	Some modest minimum specifications must be met. A data connection to the MultiVOIP unit (whether via serial port, USB2.0-with-adaptor, or web-based) must be made.
● **Hookup**	Connect power, phone, and data cables per diagram.
● **Software Installation**	This is the configuration program. It's a standard Windows software installation.
● **Phone/IP Starter Configuration**	You will enter phone numbers and IP addresses. You'll use default parameter values where possible to get the system running quickly. Use "Config Info CheckList" (page 24).
● **Phonebook Starter Configuration**	The phonebook is where you specify how calls will be routed. To get the system running quickly, you'll make phonebooks for just two voip sites.
● **Connectivity Test**	You'll find out if your voip system can carry phone calls between two sites. That means you're up and running!
● **Troubleshooting**	Detect and remedy any problems that might have prevented connectivity.

Phone/IP Details *Absolutely Needed* Before Starting the Installation

The MultiVOIP will interface with both the IP network and the phone system. You must gather information about the IP network and about the phone system so that the MultiVOIP can be configured to operate with them properly. A summary of this configuration information appears on page 24 ("Config Info CheckList").

Gather IP Information

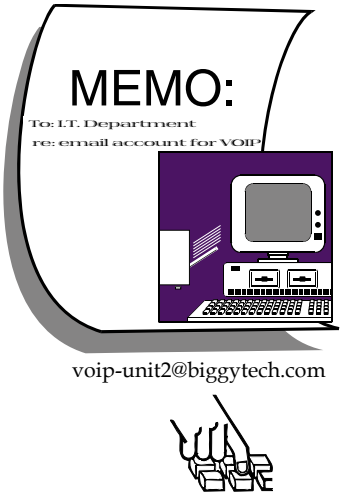
➳	*Ask your computer network administrator.*	*Info needed to operate:* **all MultiVOIP models.**
	IP Network Parameters: Record for each VOIP Site in System	
	• IP Address	
	• IP Mask	
	• Gateway	
	• Domain Name Server (DNS) Info (optional)	
	• Determine whether or not 802.1p Packet Prioritization will be used.	

Gather Telephone Information

➳	**ISDN-BRI Phone Parameters** *Ask phone company or telecom manager.*	*Needed for:* **MVP810ST MVP410ST**
	ISDN-BRI Telephony Interface Parameters: Record them for this VOIP Site	
	• In which country is this voip installed?	
	• Which operator (switch type) is used?	
	• What type of line coding use required, A-law or u-law?	
	• Determine which BRI ports will be network side and which BRI ports will be terminal side.	
	• If you are connecting the MultiVOIP to network equipment with a "U" interface, an NT1 device must be connected between them.	

Phone/IP Details Often Needed/Wanted

Obtain Email Address for VOIP (for email call log reporting)

required if log reports of VOIP call traffic are to be sent by email	**Optional**
SMTP Parameters Preparation Task: Ask Mail Server administrator to set up email account (with password) for the MultiVOIP unit itself. Be sure to give a unique identifier to each individual MultiVOIP unit. Get the IP address of the mail server computer, as well.	

Config Info CheckList

Type of Config Info Gathered	MultiVOIP Configuration screen on which to enter Config Info	Info Obtained √	Info Entered
IP info for voip unit ● IP address ● Gateway ● DNS IP (if used) ● 802.1p Prioritization (if used)	Ethernet/IP Parameters		
ISDN Layer 1 Interface (Choices: Network, Terminal)	ISDN BRI Parameters		
Clock Master Status (only a channel designated as Terminal can be used as the Clock Master)	ISDN BRI Parameters		
Switch Info ● Country ● Operator ● TEI Assignment ● A-Law or Mu-Law?	ISDN BRI Parameters		
Calling Party Number Type	ISDN BRI Parameters		
Called Party Number Type & Plan	ISDN BRI Parameters		
Country Code	Regional Parameters		
Email address for voip (optional)	SMTP Parameters		
Reminder: *Be sure to* **Save Setup** *after entering configuration values.*			

Identify Remote VOIP Site to Call

When you're done installing the MultiVOIP, you'll want to confirm that it is configured and operating properly. To do so, it's good to have another voip that you can call for testing purposes. You'll want to confirm end-to-end connectivity. You'll need IP and telephone information about that remote site.

If this is the very first voip in the system, you'll want to coordinate the installation of this MultiVOIP with an installation of another unit at a remote site.

Identify VOIP Protocol to be Used

Will you use H.323, SIP, or SPP? Each has advantages and disadvantages. Although it is possible to mix protocols in a single VOIP system, it is highly desirable to use the same VOIP protocol for all VOIP units in the system. SPP is a non-standard protocol developed by Multi-Tech. SPP is not compatible with the "Proprietary" protocol used in Multi-Tech's earlier generation of voip gateways.

Placement

Mount your MultiVOIP in a safe and convenient location where cables for your network and phone system are accessible. Rack-mounting instructions are in *Chapter 3: Mechanical Installation & Cabling* of the User Guide.

Command/Control Computer Setup (Specs & Settings)

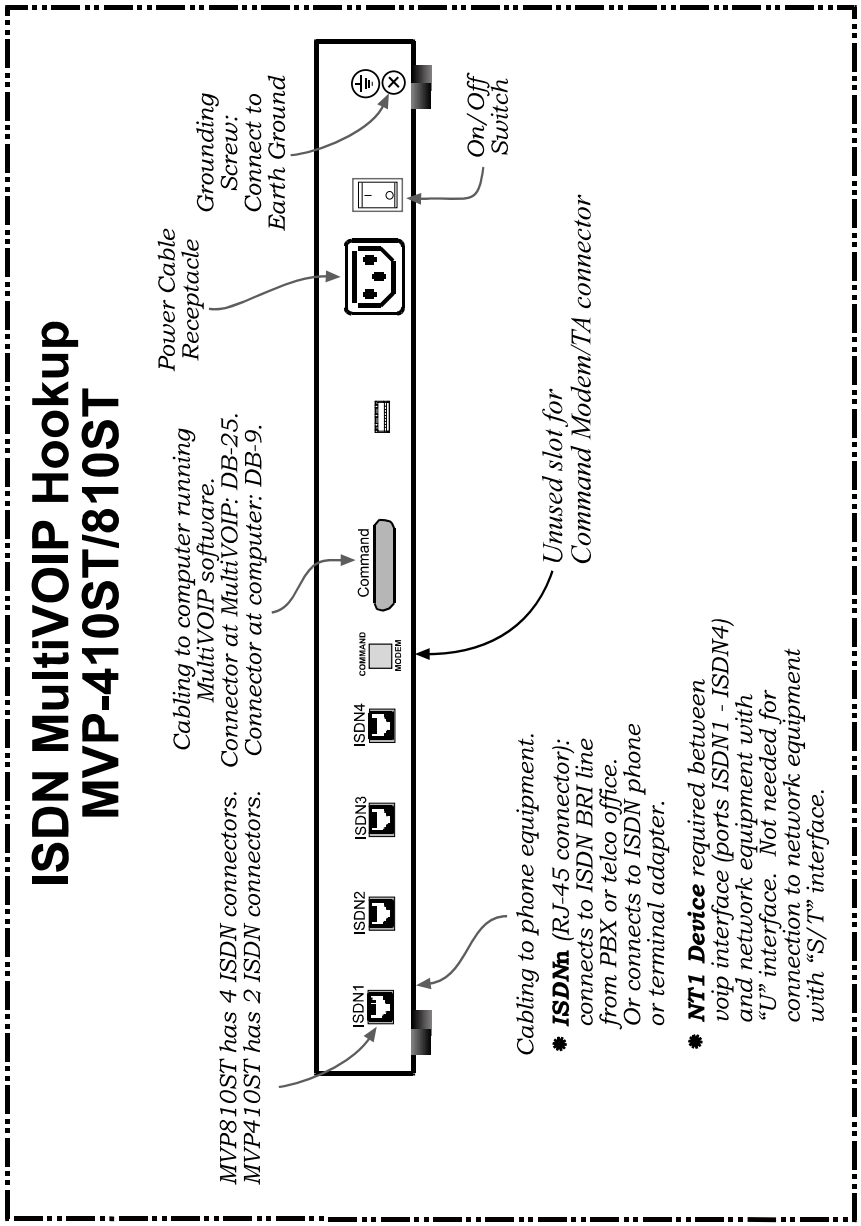
The computer used for command and control of the MultiVOIP
(a) must be an IBM-compatible PC,
(b) must use a Microsoft operating system,
(c) must be connected to your local network (Ethernet) system, and
(d) must have an available serial COM port.

The configuration tasks and control tasks the PC will have to do with the MultiVOIP are not especially demanding. Still, we recommend using a reasonably new computer. The computer that you use to configure your MultiVOIP need not be dedicated to the MultiVOIP after installation is complete.

COM port on controller PC. You'll need an available COM port on the controller PC. You'll need to know which COM port is available for use with the MultiVOIP (COM1, COM2, etc.). If your command computer has only USB2.0 ports and no serial ports, you will need a USB2.0-to-serial adaptor.

Work-Around Allowing Use of Web GUI Only. The default IP address of the MultiVOIP unit as shipped from the factory is 192.168.3.143. By installing the up-to-date Java program from the MultiVOIP CD, temporarily resetting the IP address of the command computer to 192.168.3.x, and connecting both to the same physical Ethernet network, the MultiVOIP unit's built-in- web GUI can be contacted and its IP address can be set as needed. If this is done, all subsequent configuration can be done using the MultiVOIP web GUI and the command computer would not need a serial COM port. Therefore, in the case of a computer lacking a serial port, no USB2.0-to-serial adaptor would be needed. However, under such an arrangement, the command computer would remain unable to use the MultiVOIP Windows GUI because contact with the Windows GUI requires either a COM port or the USB2.0-to-serial adaptor arrangement. (If you use this work-around, you must be sure to re-set your command PC to its original IP address afterwards.)

Hookup for MVP410ST & MVP810ST


ISDN MultiVOIP Hookup
MVP-410ST/810ST
Power Cable Receptacle
Grounding Screw: Connect to Earth Ground
On/Off Switch
Cabling to computer running MultiVOIP software. Connector at MultiVOIP: DB-25. Connector at computer: DB-9.
MVP810ST has 4 ISDN connectors. MVP410ST has 2 ISDN connectors.
Unused slot for Command Modem/TA connector
Cabling to phone equipment.
ISDN1
ISDN2
ISDN3
ISDN4
COMMAND
MODEM
Command
* ISDNn (RJ-45 connector): connects to ISDN BRI line from PBX or telco office. Or connects to ISDN phone or terminal adapter.
* NT1 Device required between voip interface (ports ISDN1 - ISDN4) and network equipment with "U" interface. Not needed for connection to network equipment with "S/T" interface.

Load MultiVOIP Control Software onto PC

For more details, see *Chapter 4: Software Installation* in this manual.

1. MultiVOIP must be properly cabled. Power must be turned on.
2. Insert MultiVOIP CD into drive. Allow 10-20 seconds for Autorun to start. If Autorun fails, go to **My Computer | CD ROM drive | Open.** Click **Autorun** icon.
3. At first dialog box, click **Install Software**.
4. At 'welcome' screen, click **Next**.
5. Follow on-screen instructions. Accept default program folder location and click **Next.**
6. Accept default icon folder location. Click **Next**. Files will be copied.
7. Select available COM port on command/control computer.
8. At completion screen, click **Finish**.
9. At the prompt "Do you want to run MultiVOIP Configuration?," click **No**. Software installation is complete.

Phone/IP Starter Configuration

For full details, see the Technical Configuration chapter of this manual.

1. Open MultiVOIP program: **Start** | **MultiVOIP** *xxx* | **Configuration**.

2. Go to **Configuration** | **Ethernet/IP**. Enter the IP parameters for your voip site. Activate Packet Prioritization (802.1p) if desired. If you use a Domain Name Server (DNS), specify its IP address. If DNS is used, you can activate the Service Record (SRV) feature. For details, see the "Technical Configuration" chapter of the *User Guide*.

3. Do you want to configure and operate the MultiVOIP unit using the web browser GUI? (It has the same functionality as the local Windows GUI, but offers remote access.)
   If NO, skip to step 5.
   If YES, continue with step 4.

4. **Web Browser GUI Setup (Optional)**. To do configuration and operation procedures using the web browser GUI, you must first set it up. To do so, follow these steps. (The browser used must be Internet Explorer 6.0 or above; or Netscape 6.0 or above; or FireFox 1.0 or above.)

A. Be sure an IP address has been assigned to the MultiVOIP unit (this must be done in the MultiVOIP Windows GUI).	**E**. Open web browser. (Note: The PC being used must be connected to and have an IP address on the same IP network that the voip is on.)
B. Save Setup in Windows GUI.	**F**. Browse to IP address of MultiVOIP unit.
C. Close the MultiVOIP Windows GUI.	**G**. If username and password have been established, enter them when prompted by voip.
D. Install Java program from MultiVOIP product CD. (Must be Java Runtime Environment 1.4.2_01 or above.) *NOTE: Required on first use of Web Browser GUI only.*	**H**. Use web browser GUI to configure or operate voip.
Need more info?	See "Web Browser Interface" in *Operation & Maintenance* chapter of **User Guide**.

Once you've begun using the web browser GUI, you can go back to the MultiVOIP Windows GUI at any time. However, you must log out of the web browser GUI before using the MultiVOIP Windows GUI.

Phone/IP Starter Configuration (continued)

5. Go to **Configuration** | **Voice/Fax**. Select **Coder** | "Automatic." At the right-hand side of the dialog box, click **OK**. If you know any specific parameter values that will apply to your system, enter them. Click **Copy Channel**. Select **Copy to All**. Click **Copy**. At main Voice/Fax Parameters screen, click **OK** to exit from the dialog box.

6. Enter telephone system information.

   Go to **Configuration** | **ISDN BRI**.
   Enter parameters obtained from phone company or PBX administrator.
   If the voip is connected to BRI extensions of a PBX or a phone company, then select "Terminal"
   in the **ISDN BRI Parameters** screen.
   If the voip is connected to ISDN terminal adapters and/or ISDN phones, then select "Network"
   in the **ISDN BRI Parameters** screen.

7. Go to **Configuration** | **Regional Parameters**. Select the **Country/Region** that fits your situation. Click **OK** and confirm. Click **OK** to exit from the dialog box.

8. Go to **Configuration** | **Regional Parameters**. In the **Country Selection for Built-In Modem** field (drop-down list), select the country that best fits your situation. (This may not be the same as your selection for the **Country/Region** field. The selections in the **Country Selection for Built-In Modem** field entail more detailed groupings of telephony parameters than do the **Country/Region** values.)

9. Do you want the phone-call logs produced by the MultiVOIP to be sent out by email (to your Voip Administrator or someone else)?
   If NO, skip to step 11.
   If YES, continue with step 10.

10. Go to **Configuration** | **SMTP**.
    SMTP lets you send phone-call log records to the Voip Administrator by email. Select **Enable SMTP**.
    You should have already obtained an email address for the MultiVOIP itself (this serves as the origination email account for email logs that the MultiVOIP can email out automatically).

    Enter this email address in the "Login Name" field.
    Type the password for this email account.

    Enter the IP address of the email server where the MultiVOIP's email account is located in the "Mail Server IP Address" field.

Typically the email log reports are sent to the Voip Administrator but they can be sent to any email address. Decide where you want the email logs sent and enter that email address in the "Recipient Address" field.

Whenever email log messages are sent out, they must have a standard Subject line. Something like "Phone Logs for Voip N" is useful. If you have more than one MultiVoip unit in the building, you'll need a unique identifier for each one (select a useful name or number for "N"). In this "Subject" field, enter a useful subject title for the log messages.

In the "Reply-To Address" field, enter the email address of your Voip Administrator.

11. Go to **Configuration** | **Logs**.
Select "Enable Console Messages."

To allow log reports by email (if desired), click **SMTP**. Click **OK**.

To do logging with a SysLog client program, click on "SysLog Server – Enable" in the **Logs** screen. To implement this function, you must install a SysLog client program. For more info, see the "SysLog Server Functions" section of the *Operation & Maintenance* chapter of the **User Guide**.

Phone/IP Starter Configuration (continued)

12. Enable premium (H.450) telephony features.

    Go to **Supplementary Services**. Select any features to be used. For Call Hold, Call Transfer, & Call Waiting, specify the key sequence that the phone user will press to invoke the feature. For Call Name Identification, specify the allowed name types to be used and a caller-id descriptor.

    If Call Forwarding is to be used, enable this feature in the Add/Edit Inbound Phone Book screen.

    After making changes, click on **OK** in the current configuration screen before moving on to the next configuration screen.

13. **RADIUS Support**. If you intend to use a RADIUS server for billing or other accounting purposes, enter the server information in the **RADIUS** screen.

14. **STUN Support**. If you are using the SIP protocol with the UDP transmission protocol, and if you want the MultiVOIP to operate behind a NAT (Network Address Translation server) using the STUN protocol (Simple Traversal of UDP through NAT), enable this feature in the **NAT Traversal** screen. You must also specify the IP address (etc.) of the STUN server you will use. The STUN server could be a local device or it could be a public STUN server accessible on the Internet.

15. **Traffic Control**. If you intend to use a traffic control server, go to **Call Signaling** and open the screen for the voip protocol you will use (**H.323**, **SIP**, or **SPP**). For H.323, you can specify a primary and alternate gatekeepers; for SIP, a primary and alternate proxy servers; for SPP, a primary and alternate registrar gateways.

    If the voip protocol is SIP, note that a separate username (number only) and password can be specified for each voip channel.

16. Go to **Save Setup | Save and Reboot**. Click OK. This will save the parameter values that you have just entered.

    The MultiVOIP's "BOOT" LED will light up while the configuration file is being saved and loaded into the MultiVOIP. Don't do anything to the MultiVOIP until the "BOOT "LED is off (a loss of power at this point could cause the MultiVOIP unit to lose the configuration settings you have made).

END OF PROCEDURE.

Phonebook Starter Configuration (*with remote voip*)

If the topic of voip phone books is new to you, it may be helpful to read the PhoneBook Tips section (page 41) before starting this procedure.

To do this part of the quick setup, you need to know of another voip that you can call to conduct a test. It should be at a remote location, typically somewhere outside of your building. You must know the phone number and IP address for that site. We are assuming here that the MultiVOIP will operate in conjunction with a PBX.

You must configure both the Outbound Phonebook and the Inbound Phonebook. A starter configuration only means that two voip locations will be set up to begin the system and establish voip communication.

Outbound Phonebook

1. Open the MultiVOIP program.
   (**Start** | **MultiVOIP xxx** | **Configuration**)
2. Go to **Phone Book** | **Outbound Phonebook** | **Add Entry**.
3. On a sheet of paper, write down the calling code of the remote voip (area code, country code, city code, etc.) that you'll be calling.

   Follow the example that best fits your situation.

North America, Long-Distance Example

Technician in Seattle (area 206) must set up one voip there, another in Chicago (area 312, downtown).

Answer: Write down **312**.

Euro, National Call Example

Technician in central London (area 0207) to set up voip there, another in Birmingham (area 0121).

Answer: write down **0121**.

Euro, International Call Example

Technician in Rotterdam (country 31; city 010) to set up one voip there, another in Bordeaux (country 33; area 05).

Answer: write down **3305**.

4. Suppose you want to call a phone number outside of your building using a phone station that is an extension from your PBX system (if present). What digits must you dial? Often a "9" or "8" must be dialed to "get an outside line" through the PBX (i.e., to connect to the PSTN). Generally, "1 "or "11" or "0" must be dialed as a prefix for calls outside of the calling code area (long-distance calls, national calls, or international calls).

   On a sheet of paper, write down the digits you must dial before you can dial a remote area code.

North America, Long-Distance Example

Seattle-Chicago system.

Seattle voip works with PBX that uses "8" for all voip calls. "1" must immediately precede area code of dialed number.

Answer: write down **81**.

Euro, National Call Example

London/Birming. system.

London voip works with PBX that uses "9" for all out-of-building calls whether by voip or by PSTN. "0" must immediately precede area code of dialed number.

Answer: write down **90**.

Euro, International Call Example

Rotterdam/Bordeaux system.
Rotterdam voip works with PBX where "9" is used for all out-of-building calls. "0" must precede all international calls.

Answer: write down **90**.

5. In the "Destination Pattern" field of the **Add/Edit Outbound Phonebook** screen, enter the digits from step 4 followed by the digits from step 3.

North America, Long-Distance Example

Seattle-Chicago system.

Answer: enter **81312** as Destination Pattern in Outbound Phone-book of Seattle voip.

Euro, National Call Example

London/Birming. system.

Leading zero of Birmingham area code is dropped when combined with national-dialing access code. (Such practices vary by country.)

Answer: enter **90121** as Destination Pattern in Outbound Phonebook of London voip. *Not 900121.*

Euro, International Call Example

Rotterdam/Bordeaux system.

Answer: enter **903305** as Destination Pattern in Outbound Phonebook of Rotterdam voip.

6. In the "Remove Prefix" field, enter the initial PBX access digit ("8" or "9").

North America, Long-Distance Example

Seattle-Chicago system.

Answer: enter **8** in "Remove Prefix" field of Seattle Outbound Phonebook.

Euro, National Call Example

London/Birming. system.

Answer: enter **9** in "Remove Prefix" field of London Outbound Phonebook.

Euro, International Call Example

Rotterdam/Bordeaux system.

Answer: enter **9** in "Remove Prefix" field of Outbound Phonebook for Rotterdam voip.

Some PBXs will not 'hand off' the "8" or "9" to the voip. But for those PBX units that do, it's important to enter the "8" or "9" in the "Remove Prefix" field in the Outbound Phonebook. This precludes the problem of having to make two inbound phonebook entries at remote voips, one to account for situations where "8" is used as the PBX access digit, and another for when "9" is used.

7. In the "Protocol Type" field group, select the voip protocol that you will use (H.323, SIP, or SPP). Use the appropriate screen under **Configuration | Call Signaling** to configure the voip protocol in detail.
8. Click **OK**. to exit from the **Add/Edit Outbound Phonebook** screen.

Inbound Phonebook

1. Open the MultiVOIP program.
   (**Start** | **MultiVOIP xxx** | **Configuration**)
2. Go to **Phone Book** | **Inbound Phonebook** | **Add Entry**.
3. In the "Remove Prefix" field, enter your local calling code (area code, country code, city code, etc.) preceded by any other "access digits" that are required to reach your local site from the remote voip location (think of it as though the call were being made through the PSTN – even though it will not be).

North America, Long-Distance Example

Seattle-Chicago system.

Seattle is area 206. Chicago employees must dial 81 before dialing any Seattle number on the voip system.

Answer: **1206** is prefix to be removed by local (Seattle) voip.

Euro, National Call Example

London/Birming. system.

Inner London is 0207 area. Birmingham employees must dial 9 before dialing any London number on the voip system.

Answer: **0207** is prefix to be removed by local (London) voip.

Euro, International Call Example

Rotterdam/Bordeaux system.
Rotterdam is country code 31, city code 010. Bordeaux employees must dial 903110 before dialing any Rotterdam number on the voip system.

Answer: **03110** is prefix to be removed by local (Rotterdam) voip.

4. In the "Add Prefix" field, enter any digits that must be dialed from your local voip to gain access to the PSTN.

North America, Long-Distance Example

Seattle-Chicago system.

On Seattle PBX, "9" is used to get an outside line.

Answer: **9** is prefix to be added by local (Seattle) voip.

Euro, National Call Example

London/Birming. system.

On London PBX, "9" is used to get an outside line.

Answer: **9** is prefix to be added by local (London) voip.

Euro, International Call Example

Rotterdam/Bordeaux system.
On Rotterdam PBX, "9" is used to get an outside line.

Answer: **9** is prefix to be added by local (Rotterdam) voip.

5. In the "Channel Number" field, enter "Hunting." A "hunting" value means the voip unit will assign the call to the first available channel. If desired, specific channels can be assigned to specific incoming calls (i.e., to any set of calls received with a particular incoming dialing pattern).

6. In the "Description" field, it is useful to describe the ultimate destination of the calls. For example, in a New York City voip system, "incoming calls to Manhattan office," might describe a phonebook entry, as might the descriptor "incoming calls to NYC local calling area." The description should make the routing of calls easy to understand. (40 characters max.)

North America, Long-Distance Example	**Euro, National Call Example**
Seattle-Chicago system. Possible Description:. Free Seattle access, all employees	London/Birming. system. Possible Description:. Local-rate London access, all empl.

Euro, International Call Example
Rotterdam/Bordeaux system. Possible Description:. Local-rate Rotterdam access, all empl.

7. Repeat steps 2-6 for each inbound phonebook entry. When all entries are complete, go to step 8.
8. Click **OK** to exit the inbound phonebook screen.
9. Click on **Save Setup**. Highlight **Save and Reboot**. Click **OK**.

Your starter inbound phonebook configuration is complete.

Phonebook Tips

Preparing the phonebook for your voip system is a complex task that, at first, seems quite daunting. These tips may make the task easier.

1. **Use Dialing Patterns, Not Complete Phone Numbers**. You will not generally enter complete phone numbers in the voip phonebook. Instead, you'll enter "destination patterns" that involve area codes and other digits. If the destination pattern is a whole area code, you'll be assigning all calls to that area code to go to a particular voip which has a unique IP address. If your destination pattern includes an area code plus a particular local phone exchange number, then the scope of calls sent through your voip system will be narrowed (only calls within that local exchange will be handled by the designated voip, not all calls in that whole area code). In general, when there are fewer digits in your destination pattern, you are asking the voip to handle calls to more destinations.

2. **The Four Types of Phonebook Digits Used. Important!**
"Destination patterns" to be entered in your phonebook will generally consist of:

(a) calling area codes,
(b) access codes,
(c) local exchange numbers, and
(d) specialized codes.

Although voip phonebook entries may look confusing at first, it's useful to remember that all the digits in any phonebook entry must be of one of these four types.

(a) **calling area codes**. There are different names for these around the world: "area codes," "city codes," "country codes," etc. These codes, are used when making non-local calls. They always precede the phone number that would be dialed when making a local call.

(b) **access codes**. There are digits (*PSTN access codes*) that must be dialed to gain access to an operator, to access the publicly switched 'long-distance' calling system(North America), to access the publicly switched 'national' calling system (Europe and elsewhere), or to access the publicly switched 'international' calling system (worldwide).

There are digits (*PBX access codes*) that must be dialed by phones connected to PBX systems or key systems. Often a "9" must be dialed on a PBX phone to gain access to the PSTN ('to get an outside line'). Sometimes "8" must be dialed on a PBX phone to divert calls onto a leased line or to a voip system. However, sometimes PBX systems are 'smart' enough to route calls to a voip system without a special access code (so that "9" might still be used for all calls outside of the building).

There are also digits (*special access codes*) that must be dialed to gain access to a particular discount long-distance carrier or to some other closed or proprietary telephone system.

(c) **local exchange numbers**. Within any calling area there will be many local exchange numbers. A single exchange may be used for an entire small town. In cities, an exchange may be used for a particular neighborhood (although exchanges in cities do not always cover easily discernible areas). Organizations like businesses, governments, schools, and universities are also commonly assigned exchange numbers for their exclusive use. In some cases, these organizational-assigned exchanges can become non-localized because the exchange is assigned to one facility and linked, by the organization's private network, to other sometimes distant locations.

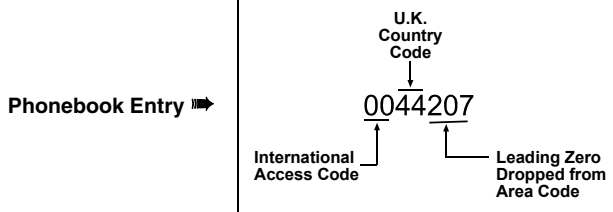
(d) **specialized codes**. Some proprietary voip units assign, to sites and phone stations, numbers that are not compatible with PSTN numbering. This can also occur in PBX or key systems. These specialized numbers must be handled on a case-by-case basis.

3. **Knowing When to Drop Digits**.

When calling area codes and access codes are used in combination, a leading "1" or "0" must sometimes be dropped.

Example

Area code for Inner London is listed as "0207." However, in international calls the leading "0" is dropped.



4. **Using a Comma**.

Commas are used in telephone dialing strings to indicate a pause to allow a dial tone to appear (common on PBX and key systems). Commas may be used only in the "Add Prefix" field of the Inbound Phonebook.

Detail

, **= 1-second pause**

in many PBX systems
(not needed in all)

5. **Ease of Use**. The phonebook setup determines how easy the voip system is to use. Generally, you'll want to make it so dialing a voip call is very similar to dialing any other number (on the PSTN or through the PBX).

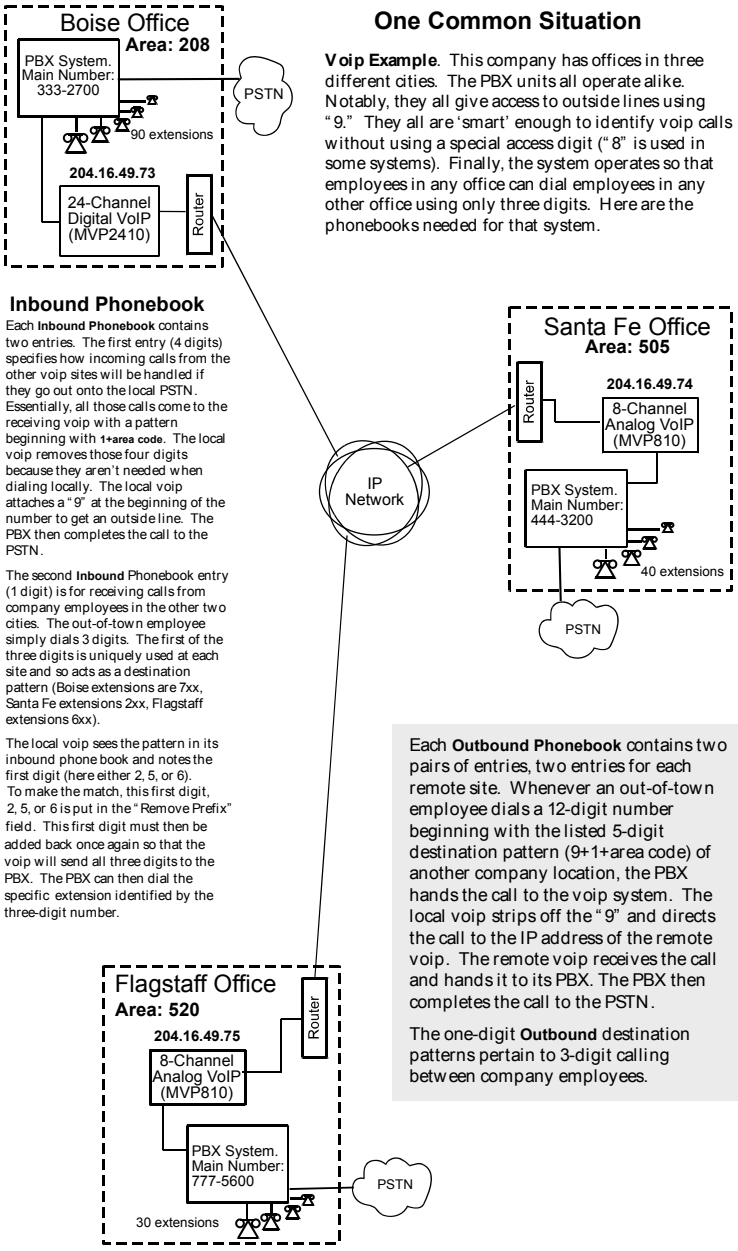
6. **Avoid Unintentional Calls to Official/Emergency Numbers**. Dialing a voip call will typically be somewhat different than ordinary dialing. Because of this, it's possible to set up situations, quite unwittingly, where phone users may be predisposed to call official numbers without intending to do so. Conversely, a voip/PBX system might also make it difficult to place an official/emergency call when one intends to do so. Study your phonebook setup and do some test-dialing on the system to avoid these pitfalls.

7. **Inbound/Outbound Pattern Matching**. In general, the Inbound Phonebook entries of the local voip unit will match the Outbound Phonebook entries of the remote voip unit. Similarly, the Outbound Phonebook entries of the local voip unit will match the Inbound Phonebook entries of the remote voip unit. There will often be non-matching entries, but it's nonetheless useful to notice the matching between the phonebooks.

8. **Simulating Network in-lab/on-benchtop**. One common method of configuring a voip network is to set up a local IP network in a lab, connect voip units to it, and perhaps have phones connected on channel banks to make test calls.

NOTES

Phonebook Example



One Common Situation

Voip Example. This company has offices in three different cities. The PBX units all operate alike. Notably, they all give access to outside lines using "9." They all are 'smart' enough to identify voip calls without using a special access digit ("8" is used in some systems). Finally, the system operates so that employees in any office can dial employees in any other office using only three digits. Here are the phonebooks needed for that system.

Inbound Phonebook

Each **Inbound Phonebook** contains two entries. The first entry (4 digits) specifies how incoming calls from the other voip sites will be handled if they go out onto the local PSTN. Essentially, all those calls come to the receiving voip with a pattern beginning with **1+area code**. The local voip removes those four digits because they aren't needed when dialing locally. The local voip attaches a "9" at the beginning of the number to get an outside line. The PBX then completes the call to the PSTN.

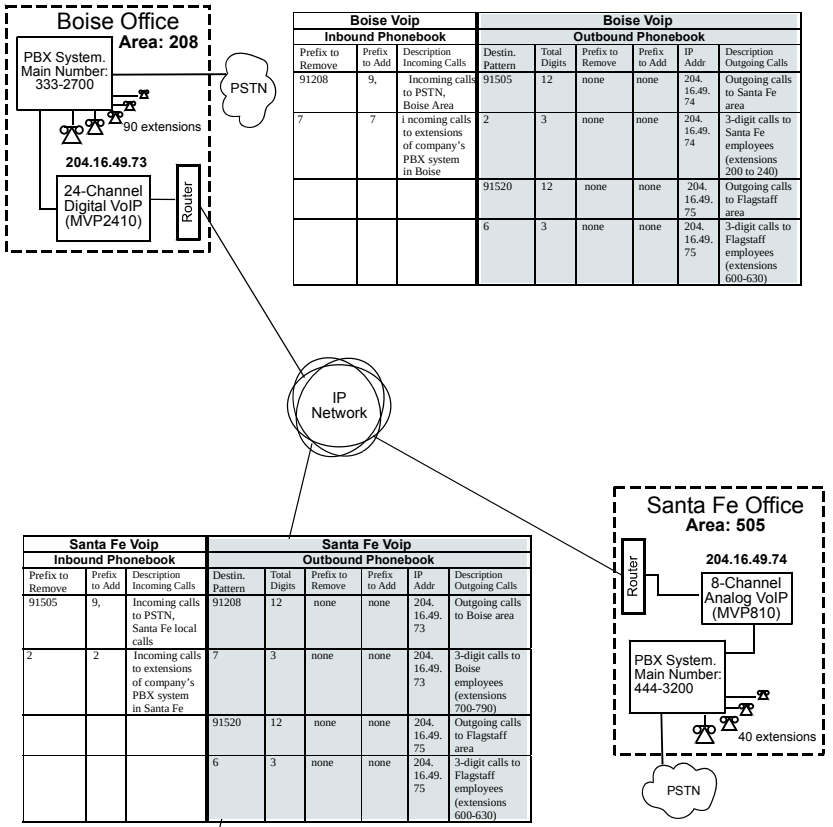
The second **Inbound** Phonebook entry (1 digit) is for receiving calls from company employees in the other two cities. The out-of-town employee simply dials 3 digits. The first of the three digits is uniquely used at each site and so acts as a destination pattern (Boise extensions are 7xx, Santa Fe extensions 2xx, Flagstaff extensions 6xx).

The local voip sees the pattern in its inbound phone book and notes the first digit (here either 2, 5, or 6). To make the match, this first digit, 2, 5, or 6 is put in the "Remove Prefix" field. This first digit must then be added back once again so that the voip will send all three digits to the PBX. The PBX can then dial the specific extension identified by the three-digit number.

Each **Outbound Phonebook** contains two pairs of entries, two entries for each remote site. Whenever an out-of-town employee dials a 12-digit number beginning with the listed 5-digit destination pattern (9+1+area code) of another company location, the PBX hands the call to the voip system. The local voip strips off the "9" and directs the call to the IP address of the remote voip. The remote voip receives the call and hands it to its PBX. The PBX then completes the call to the PSTN.

The one-digit **Outbound** destination patterns pertain to 3-digit calling between company employees.

Voip Sites with Phonebooks



Boise Voip			Boise Voip					
Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls
91208	9,	Incoming calls to PSTN, Boise Area	91505	12	none	none	204. 16.49. 74	Outgoing calls to Santa Fe area
7	7	i ncoming calls to extensions of company's PBX system in Boise	2	3	none	none	204. 16.49. 74	3-digit calls to Santa Fe employees (extensions 200 to 240)
			91520	12	none	none	204. 16.49. 75	Outgoing calls to Flagstaff area
			6	3	none	none	204. 16.49. 75	3-digit calls to Flagstaff employees (extensions 600-630)

Santa Fe Voip			Santa Fe Voip					
Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls
91505	9,	Incoming calls to PSTN, Santa Fe local calls	91208	12	none	none	204. 16.49. 73	Outgoing calls to Boise area
2	2	Incoming calls to extensions of company's PBX system in Santa Fe	7	3	none	none	204. 16.49. 73	3-digit calls to Boise employees (extensions 700-790)
			91520	12	none	none	204. 16.49. 75	Outgoing calls to Flagstaff area
			6	3	none	none	204. 16.49. 75	3-digit calls to Flagstaff employees (extensions 600-630)

Flagstaff Voip			Flagstaff Voip					
Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls
91520	9	Incoming calls to PSTN, Flagstaff local calls	91505	12	none	none	204.16 .49.74	Outgoing calls to Santa Fe area
6	6	Incoming calls to extensions of company's PBX system in Flagstaff	2	3	none	none	204.16 .49.74	3-digit calls to Santa Fe employees (extensions 200-240)
			91208	12	none	none	204.16 .49.73	Outgoing calls to Boise area
			7	3	none	none	204.16 .49.73	3-digit calls to Boise employees (extensions 700-790)

Sample Phonebooks Enlarged

Boise Voip			Boise Voip					
Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls
91208	9,	Incoming calls to PSTN, Boise Area	91505	12	none	none	204. 16.49. 74	Outgoing calls to Santa Fe area
7	7	i ncoming calls to extensions of company's PBX system in Boise	2	3	none	none	204. 16.49. 74	3-digit calls to Santa Fe employees (extensions 200 to 240)
			91520	12	none	none	204. 16.49. 75	Outgoing calls to Flagstaff area
			6	3	none	none	204. 16.49. 75	3-digit calls to Flagstaff employees (extensions 600-630)

Santa Fe Voip			Santa Fe Voip					
Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls
91505	9,	Incoming calls to PSTN, Santa Fe local calls	91208	12	none	none	204. 16.49. 73	Outgoing calls to Boise area
2	2	Incoming calls to extensions of company's PBX system in Santa Fe	7	3	none	none	204. 16.49. 73	3-digit calls to Boise employees (extensions 700-790)
			91520	12	none	none	204. 16.49. 75	Outgoing calls to Flagstaff area
			6	3	none	none	204. 16.49. 75	3-digit calls to Flagstaff employees (extensions 600-630)

Flagstaff Voip			Flagstaff Voip					
Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls
91520	9	Incoming calls to PSTN, Flagstaff local calls	91505	12	none	none	204.16 .49.74	Outgoing calls to Santa Fe area
6	6	Incoming calls to extensions of company's PBX system in Flagstaff	2	3	none	none	204.16 .49.74	3-digit calls to Santa Fe employees (extensions 200-240)
			91208	12	none	none	204.16 .49.73	Outgoing calls to Boise area
			7	3	none	none	204.16 .49.73	3-digit calls to Boise employees (extensions 700-790)

Phonebook Worksheet

Voip Location/ID:_____

Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls

Other Details:

Voip Location/ID:_____

Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls

Other Details:

Voip Location/ID:_____

Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls

Other Details:

Enlarged Phonebook Worksheet

Voip Location/ID:

Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls

Other Details:

Connectivity Test

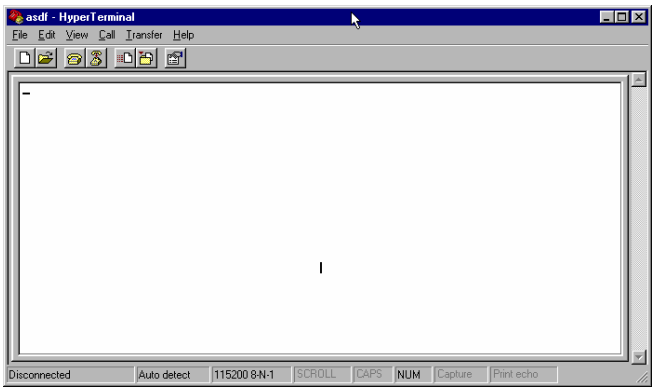
The procedures "Phone/IP Starter Configuration" and "Phonebook Starter Configuration" must be completed before you can do this procedure.

1. These connections must be made:
   - MultiVOIP to local PBX or telco
   - MultiVOIP to command PC
   - MultiVOIP to Internet

2. Inbound Phonebook and Outbound Phonebook must both be set up with at least one entry in each. These entries must allow for connection between two voip units.
3. Console messages must be enabled. (If this has not been done already, go, in the MultiVOIP GUI, to Configuration | Logs and select the "Console Messages" checkbox.

4. You now need to free up the COM port connection (currently being used by the MultiVOIP program) so that the HyperTerminal program can use it. To do this, you can either (a) click on **Connection** in the sidebar and select "Disconnect" from the drop-down box, or (b) close down the MultiVOIP program altogether.

5. Open the **HyperTerminal** program.



6. Use HyperTerminal to receive and record console messages from the MultiVOIP unit. To do so, set up HyperTerminal as follows (setup shown is for Windows NT4; details will differ slightly in other MS operating systems):

   - In the upper toolbar of the HyperTerminal screen, click on the **Properties** button.
   - In the "Connect To" tab of the **Connection Properties** dialog box, click on the **Configure** button.
   - In the next dialog box, on the "General" tab, set "Maximum Speed" to 115200 bps.
   - On the "Connection" tab, set connection preferences to:

     Data bits: 8

     Parity: none

     Stop bits: 1

   - Click **OK** twice to exit settings dialog boxes.

7. Make VOIP call.

for ISDN-BRI MultiVOIPs (MVP-410ST/810ST)
in terminal mode;
Make call from an extension of the local PBX to the voip extension.

8. Read console messages recorded on HyperTerminal.

Console Messages from **Originating VOIP**. The voip unit that originates the call will send back messages like that shown below.

```
[00026975]  CAS[0] : RX : ABCD = 1, 1, 1, 1,Pstn State[1]
            TimeStamp : 26975
[00027190]  CAS[0] : TX : ABCD = 1, 1, 1, 1
[00027190]  PSTN: cas seizure detected on 0
[00027440]  CAS[0] : TX : ABCD = 0, 0, 0, 0
[00033290]  PSTN:call detected on 0 num=17637175662*
[00033290] H323IF[0]:destAddr =
            TA:200.2.10.5:1720,NAME:Mounds
            View,TEL:17637175662,17637175662
 [00033290]  H323IF[0]:srcAddr  = NAME:New
            York,TA:200.2.9.20
 [00033440]  H323IF [0]:cmCallStateProceeding
[00033500]  H323[0]: Remote Information (Q931): MultiVOIP
            - T1
[00033565]  CAS[0] : TX : ABCD = 1, 1, 1, 1
[00033675]  H323IF [0]: MasterSlaveStatus=Slave
[00033675]  H323IF[0]:FastStart Setup Not Used
[00033690]  CAS[0] : TX : ABCD = 1, 1, 1, 1
[00033755]  H323IF[0]: Coder used 'g7231'
[00033810]  PSTN:pstn call connected on 0
```

Console Messages from **Terminating VOIP**. The voip unit connected to the phone where the call is answered will send back messages like that shown below.

```
[00170860]  H323[0]: New incoming call
[00170860]  PSTNIF : Placing call on channel 0  Outbound
             digit 7175662
[00170885]  CAS[0] : TX : ABCD = 1, 1, 1, 1
[00171095]  H323IF [0]: MasterSlaveStatus=Master
[00171105]  CAS[0] : RX : ABCD = 1, 1, 1, 1,Pstn State[7]
             TimeStamp : 171105
[00171105]  H323IF[0]: Coder used 'g7231'
[00171110]  H323IF[0]:FastStart Setup Not Used
[00171110]  H323IF[0]: Already opened the outgoing logical
             channel
[00171110]  H323IF[0]: Coder used 'g7231'
[00171315]  CAS[0] : RX : ABCD = 0, 0, 0, 0,Pstn State[9]
             TimeStamp : 171315
[00172275]  PSTN: dialing digit ended on 0
[00172285]  PSTN: pstn proceeding indication on 0
[00172995]  CAS[0] : RX : ABCD = 1, 1, 1, 1,Pstn State[12]
             TimeStamp : 172995
[00173660]  CAS[0] : TX : ABCD = 1, 1, 1, 1
[00173760]  PSTN:pstn call connected on 0
```

9. When you see the following message, end-to-end voip connectivity has been achieved.

   "**PSTN: pstn call connected on X**"

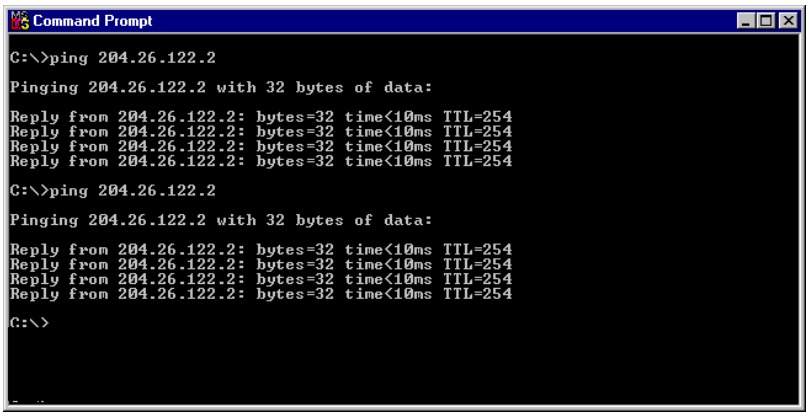
   where x is the number of the voip channel carrying the call

10. If the HyperTerminal messages do not confirm connectivity, go to the ***Troubleshooting*** procedure below.

Troubleshooting

If you cannot establish connectivity between two voips in the system, follow the steps below to determine the problem.

1. Ping both MultiVOIP units to confirm connectivity to the network.


Command Prompt
C:\>ping 204.26.122.2
Pinging 204.26.122.2 with 32 bytes of data:
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
C:\>ping 204.26.122.2
Pinging 204.26.122.2 with 32 bytes of data:
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
C:\>

2. Verify the telephone connections.
   - Check cabling. Are connections well seated? To correct receptacle?
   - If terminal equipment is connected to the voip, then "Network" should be selected for that BRI interface in the **ISDN BRI Parameters** screen.
     Note: *Each BRI interface is separately configurable.*
   - If network equipment such as an ISDN BRI PBX or an ISDN BRI line from a phone company is connected to the voip, then "Terminal" should be selected for that BRI interface in the **ISDN BRI Parameters** screen.
   - Was the proper country and operator chosen?
     Was the proper type of line coding (A-law or u-law) chosen?
3. Verify phonebook configuration.
4. Observe console messages while placing a call. Look for error messages indi-cating phonebook problems, network problems, voice-coder mismatches, etc.

Chapter 3: Mechanical Installation and Cabling

Introduction

When the MVP410ST or MVP810ST MultiVOIP units are to be installed into a rack, two able-bodied persons should participate.

Please read the safety notices before beginning installation.

Safety Warnings

Lithium Battery Caution

A lithium battery on the voice/fax channel board provides backup power for the timekeeping capability. The battery has an estimated life expectancy of ten years.

When the battery starts to weaken, the date and time may be incorrect. If the battery fails, the board must be sent back to Multi-Tech Systems for battery replacement.

Warning: There is danger of explosion if the battery is incorrectly replaced.

Safety Warnings Telecom

1. Never install telephone wiring during a lightning storm.
2. This product must be disconnected from power source and telephone network interface when servicing.
3. This product is to be used with UL and UL listed computers.
4. Never touch uninsulated telephone wires or terminals unless the telephone line has been disconnected at the network interface.
5. Use caution when installing or modifying telephone lines.
6. Avoid using a telephone (other than a cordless type) during an electrical storm. There may be a remote risk of electrical shock from lightning.
7. Do not use a telephone in the vicinity of a gas leak – not even to report a gas leak.
8. To reduce the risk of fire, use only a UL-listed 26 AWG or larger telecommunication line cord.
9. Never install a telephone jack in wet locations unless the jack is specifically designed for wet locations.

Unpacking Your MVP-410ST/810ST MultiVOIP

When unpacking your MultiVOIP, check to see that all of the items shown in Figure 3-1 are included in the box. If any box contents are missing, contact MultiTech Tech Support at 1-800-972-2439.

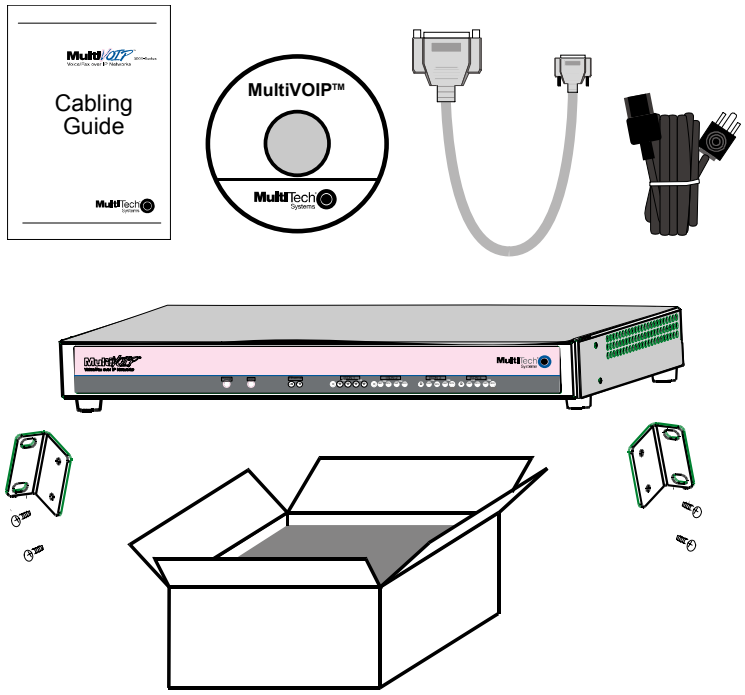


Figure 3-1: Unpacking the MVP-410ST/810ST

Rack Mounting Instructions for MVP410ST & MVP810ST

The MultiVOIPs can be mounted in an industry-standard EIA 19-inch rack enclosure, as shown in Figure 3-2.

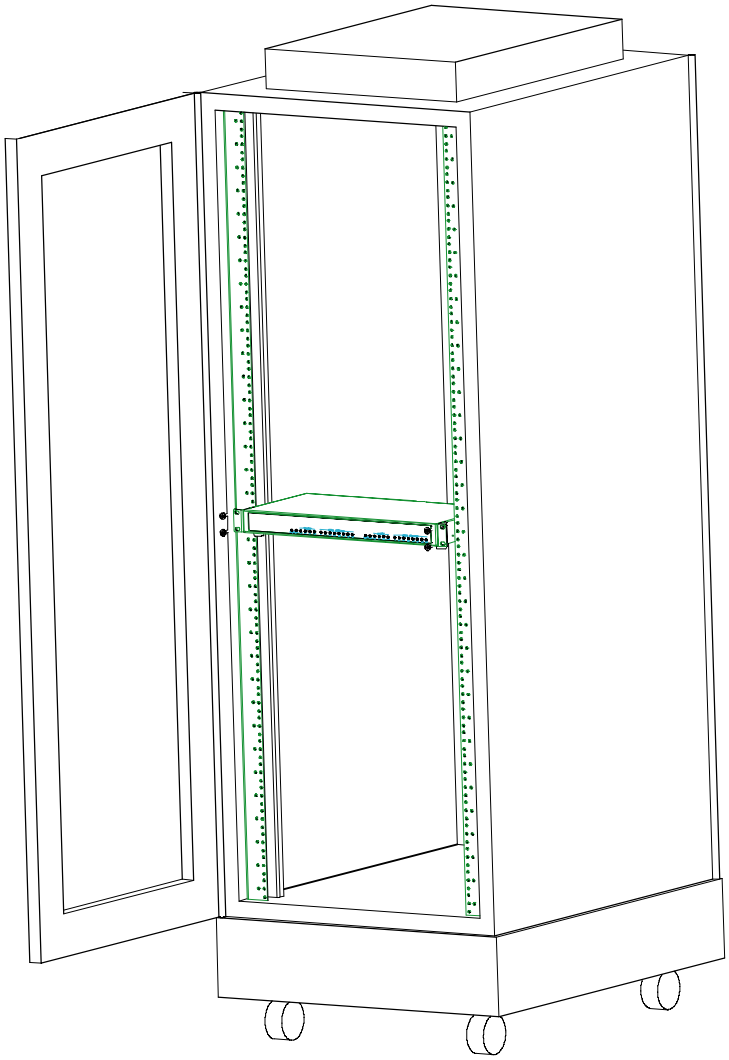


Figure 3-2: Rack-Mounting (MVP410ST or MVP810ST)

Safety Recommendations for Rack Installations

Ensure proper installation of the unit in a closed or multi-unit enclosure by following the recommended installation as defined by the enclosure manufacturer. Do not place the unit directly on top of other equipment or place other equipment directly on top of the unit. If installing the unit in a closed or multi-unit enclosure, ensure adequate airflow within the rack so that the maximum recommended ambient temperature is not exceeded. Ensure that the unit is properly connected to earth ground by verifying that it is reliably grounded when mounted within a rack. If a power strip is used, ensure that the power strip provides adequate grounding of the attached apparatus.

When mounting the equipment in the rack, make sure mechanical loading is even to avoid a hazardous condition, such as loading heavy equipment in rack unevenly. The rack used should safely support the combined weight of all the equipment it supports.

Ensure that the mains supply circuit is capable of handling the load of the equipment. See the power label on the equipment for load requirements (full specifications for MultiVOIP models are presented in chapter 1 of this manual).

Maximum ambient temperature for the unit is 60 degrees Celsius (140 degrees Fahrenheit) at 20-90% non-condensing relative humidity. This equipment should only be installed by properly qualified service personnel. Only connect like circuits. In other words, connect SELV (Secondary Extra Low Voltage) circuits to SELV circuits and TN (Telecommunications Network) circuits to TN circuits.

19-Inch Rack Enclosure Mounting Procedure

Attaching the MultiVOIP to a rack-rail of an EIA 19-inch rack enclosure will certainly require two persons. Essentially, the technicians must attach the brackets to the MultiVOIP chassis with the screws provided, as shown in Figure 3-3, and then secure unit to rack rails by the brackets, as shown in Figure 3-4. Because equipment racks vary, screws for rack-rail mounting are not provided. Follow the instructions of the rack manufacturer and use screws that fit.

1. Position the right rack-mounting bracket on the MultiVOIP using the two vertical mounting screw holes.
2. Secure the bracket to the MultiVOIP using the two screws provided.
3. Position the left rack-mounting bracket on the MultiVOIP using the two vertical mounting screw holes.
4. Secure the bracket to the MultiVOIP using the two screws provided.
5. Remove feet (4) from the MultiVOIP unit.
6. Mount the MultiVOIP in the rack enclosure per the rack manufacture's mounting procedure.

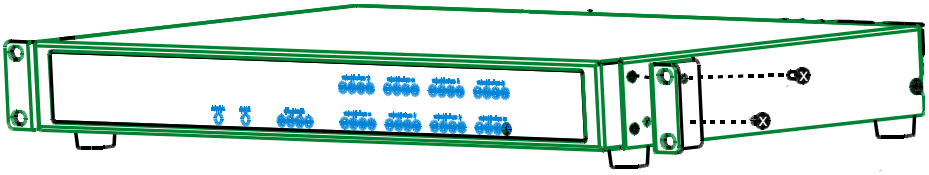


Figure 3-3: Bracket Attachment for Rack Mounting (MVP410ST & MVP810ST)

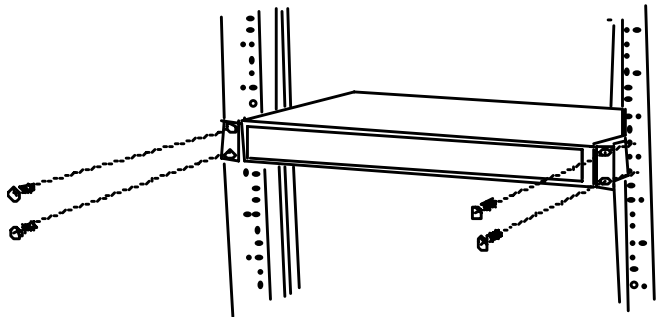


Figure 3-4: Attaching MultiVOIP to Rack Rail (MVP410ST & MVP810ST)

Cabling Procedure for MVP-410ST/810ST

Cabling involves connecting the MultiVOIP to your LAN and telephone equipment.

1. Connect the power cord supplied with your MultiVOIP to a live AC outlet and to the power connector on the back of the MultiVOIP as shown at top right in Figure 3-5.

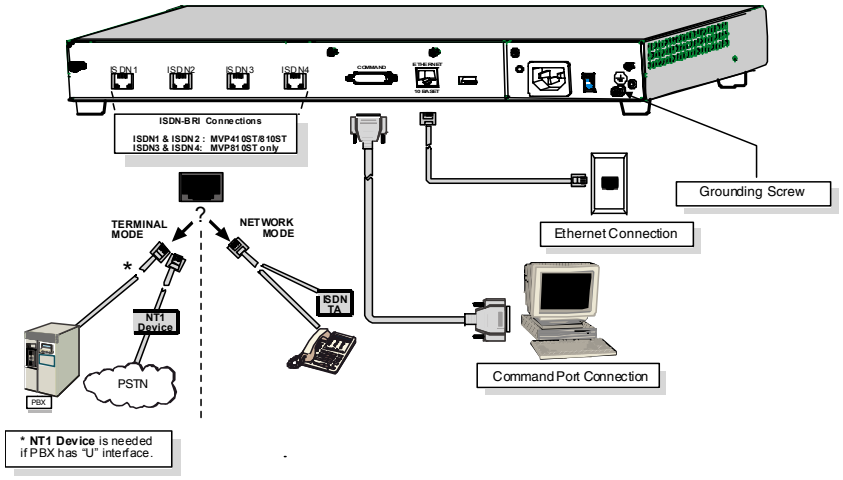
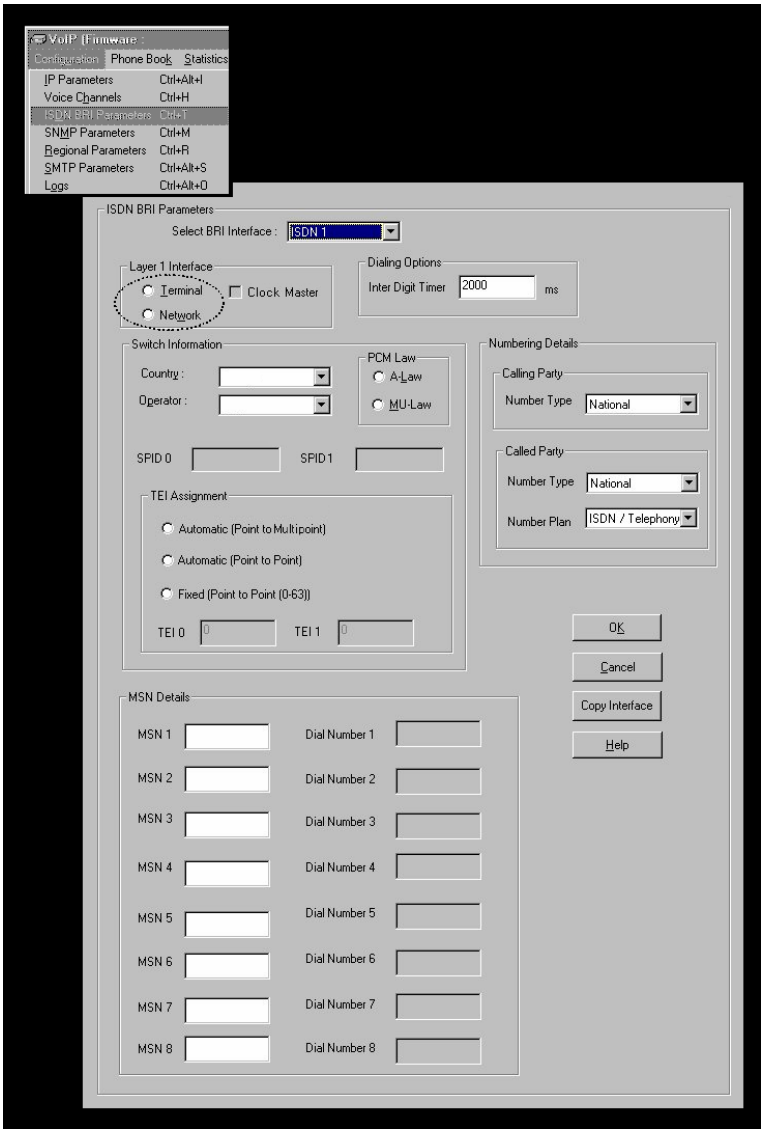


Figure 3-5: Cabling for MVP-410ST/810ST

2. Connect the MultiVOIP to a PC by using a DB-25 (male) to DB-9 (female) cable. Plug the DB-25 end of the cable into the Command port of the MultiVOIP and the other end into the PC serial port. See Figure 3-5. If your computer has only USB2.0 ports and no serial ports, you will need a serial-to-USB2.0 adapter.
3. Connect a network cable to the **ETHERNET 10BASET** connector on the back of the MultiVOIP. Connect the other end of the cable to your network.

4. **Terminal Mode**. When a voip ISDN connector is to be connected to a PBX extension line or to a telco line, select "Terminal" as the "Layer 1 Interface" in the **ISDN Parameters** screen. When making cable connections, an NT1 device will be needed between the MultiVOIP and the PSTN or between the MultiVOIP and any PBX with a "U" interface. (For more information, see *Appendix B: Cable Pinouts* in this manual.) Connect cables between voip ISDN connectors and network equipment.

   NOTE: In order to operate in **Terminal** mode, the network equipment to which you will be connecting (e.g., PBX) must support D-channel signaling in its ISDN-S/T interface.



Network Mode. When a voip ISDN connector is to be connected to an ISDN phone station or to an ISDN terminal adapter (TA), select "Network" as the "Layer 1 Interface" in the **ISDN Parameters** screen of the MultiVOIP software. Connect cables between voip ISDN connectors and phone or TA.

NOTE. Any ISDN phone stations connected to the MVP-410ST/810ST must provide their own operating power. That is, the MVP-410ST/810ST does not supply power for ISDN phone stations.

5. Repeat the above step to connect the remaining ISDN telephone equipment to each ISDN connector on your MultiVOIP. Be aware that you can assign each ISDN line separately and independently to either Network mode or Terminal mode. That is, all ISDN lines do not have to be assigned in to the same operating mode.

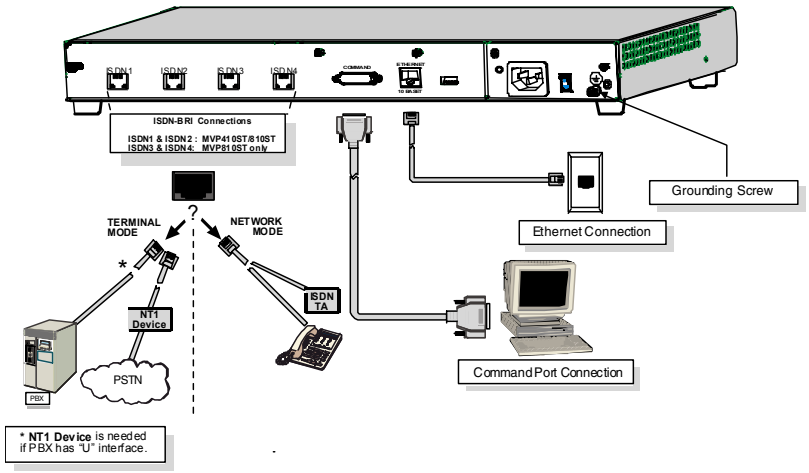


Figure 3-6: ISDN/BRI Voip Connections for GND & Remote Configuration Modem

6. Ensure that the unit is properly connected to earth ground by verifying that it is reliably grounded when mounted within a rack.

   This can be accomplished by connecting a grounding wire between the chassis grounding screw (see Figure 3-6) and a metallic object that will provide an electrical ground.

7. Ensure that the MultiVOIP unit is properly connected to earth ground.

   This can be accomplished by connecting a grounding wire between the chassis grounding screw (see Figure 3-7) and a connector securely connected to the ground (GND) of the building's electrical system.

   Ground Connection Details. Ensure that the unit is securely and permanently connected to an earth ground (GND) with a ground wire of 18 gauge (18 AWG) or thicker. The ground wire needs to be installed between the grounding screw on the MultiVOIP chassis and a permanent earth ground. Whether the MultiVOIP unit is used in a rack or on a desktop, you must verify that the earth ground connection is permanent and reliable. In order for the ground

connection to be considered permanent, the grounding wire must connect to the earth ground of the building's electrical wiring system and the ground connection must use a screw terminal or other reliable means of fastening. The ground connection must not be as easily disconnected as, for example, a power cord.

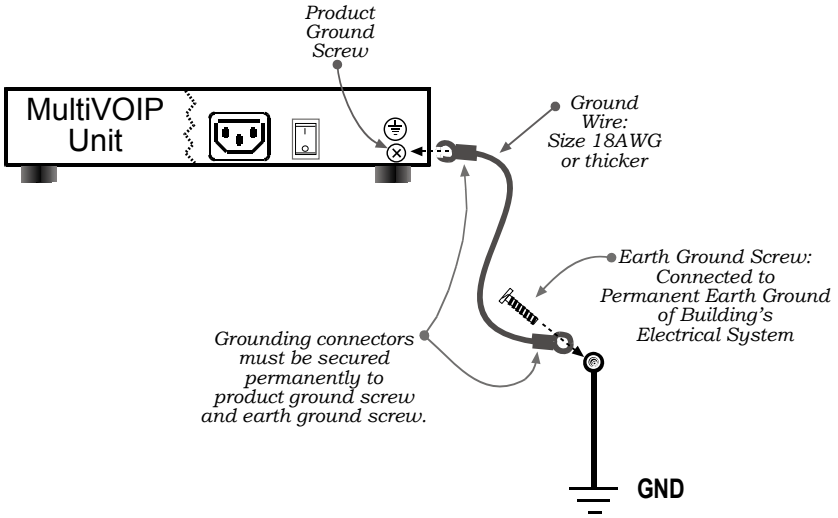

Product Ground Screw
MultiVOIP Unit
Ground Wire: Size 18AWG or thicker
Earth Ground Screw: Connected to Permanent Earth Ground of Building's Electrical System
Grounding connectors must be secured permanently to product ground screw and earth ground screw.
GND

Figure 3-7. Earth Ground Details

8. Turn on power to the MultiVOIP by placing the ON/OFF switch on the back panel to the ON position. Wait for the **Boot** LED on the MultiVOIP to go off before proceeding. This may take a few minutes.

   Proceed to Chapter 4 to load the MultiVOIP software.

Proceed to Chapter 4 to load the MultiVOIP software.

Chapter 4: Software Installation

Introduction

Configuring software for your MultiVOIP entails three tasks:
(1) loading the software onto the PC (this is "Software Installation and is discussed in this chapter),
(2) setting values for telephony and IP parameters that will fit your system (this is "Technical Configuration" and it is discussed in Chapter 5), and
(3) establishing "phonebooks" that contain the various dialing patterns for VOIP calls made to different locations (this is "Phonebook Configuration" and it is discussed in Chapter 6 for North American (T1) telephony standards and in Chapter 7 for European (E1) telephony standards.

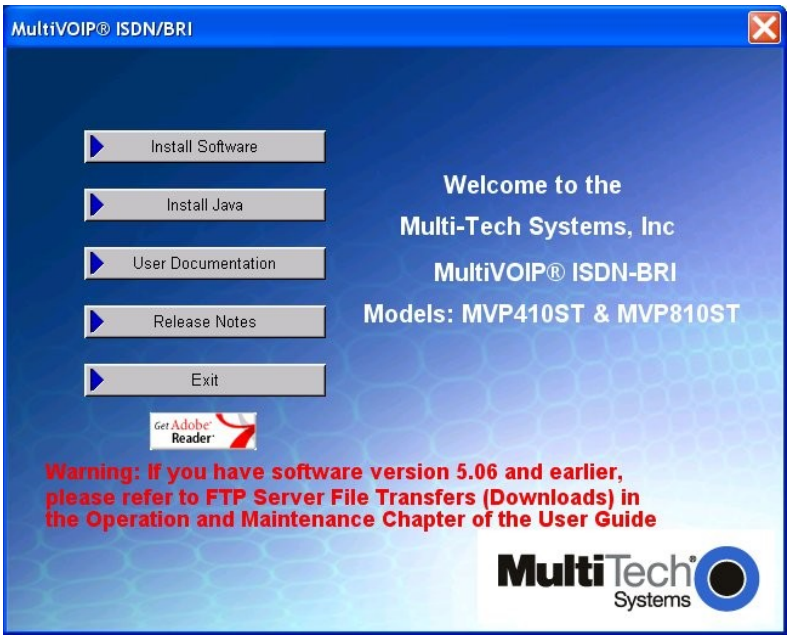
Loading MultiVOIP Software onto the PC

The software loading procedure does not present every screen or option in the loading process. It is assumed that someone with a thorough knowledge of Windows and the software loading process is performing the installation.

The MultiVOIP software and User Guide are contained on the MultiVOIP product CD. Because the CD is auto-detectable, it will start up automatically when you insert it into your CD-ROM drive. When you have finished loading your MultiVOIP software, you can view and print the User Guide by clicking on the **View Manuals** icon.

1. Be sure that your MultiVOIP has been properly cabled and that the power is turned on.

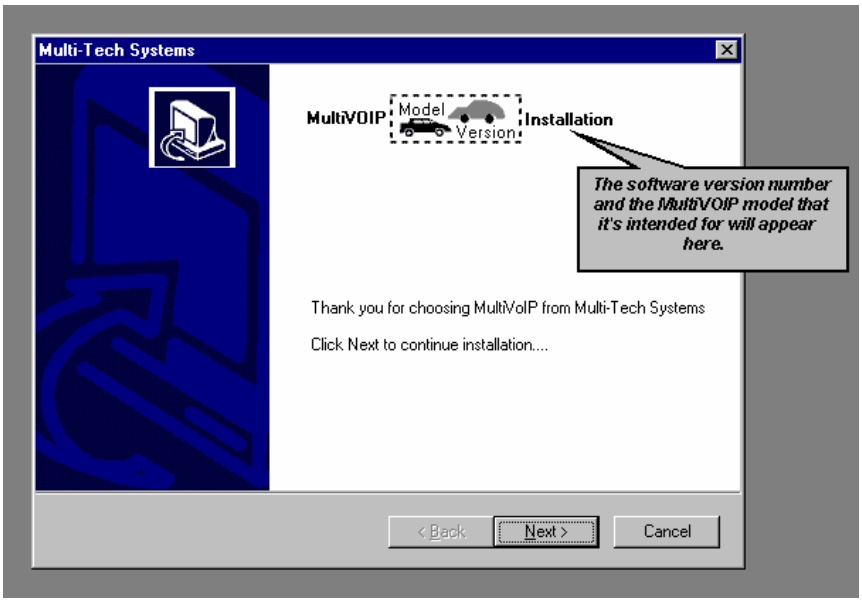
2. Insert the MultiVOIP CD into your CD-ROM drive. The CD should start automatically. It may take 10 to 20 seconds for the Multi-Tech CD installation window to display.


MultiVOIP® ISDN/BRI
Install Software
Install Java
User Documentation
Release Notes
Exit
Get Adobe Reader
Welcome to the
Multi-Tech Systems, Inc
MultiVOIP® ISDN-BRI
Models: MVP410ST & MVP810ST
Warning: If you have software version 5.06 and earlier, please refer to FTP Server File Transfers (Downloads) in the Operation and Maintenance Chapter of the User Guide
MultiTech Systems

If the Multi-Tech Installation CD window does not display automatically, click **My Computer**, then right click the **CD ROM drive** icon, click **Open**, and then click the **Autorun** icon.

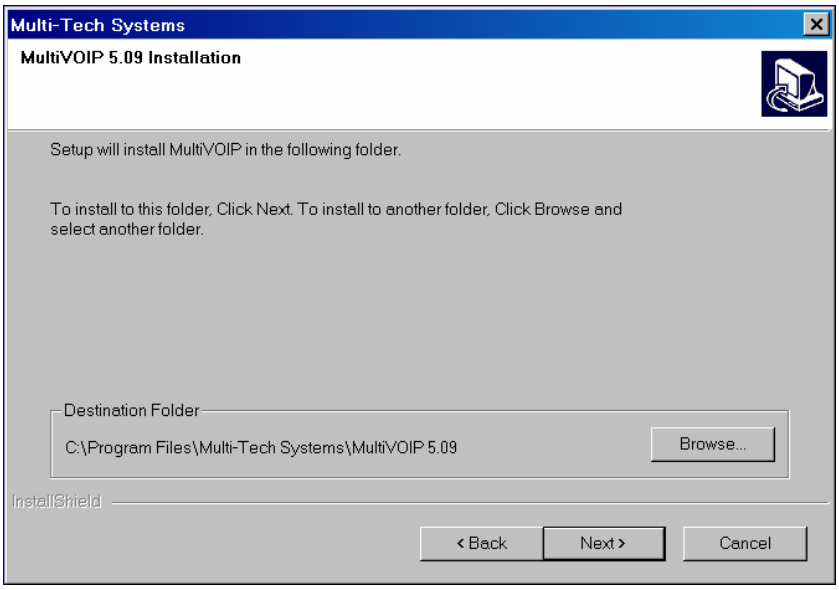
3. When the Multi-Tech Installation CD dialog box appears, click the **Install Software** icon.

4. A 'welcome' screen appears.



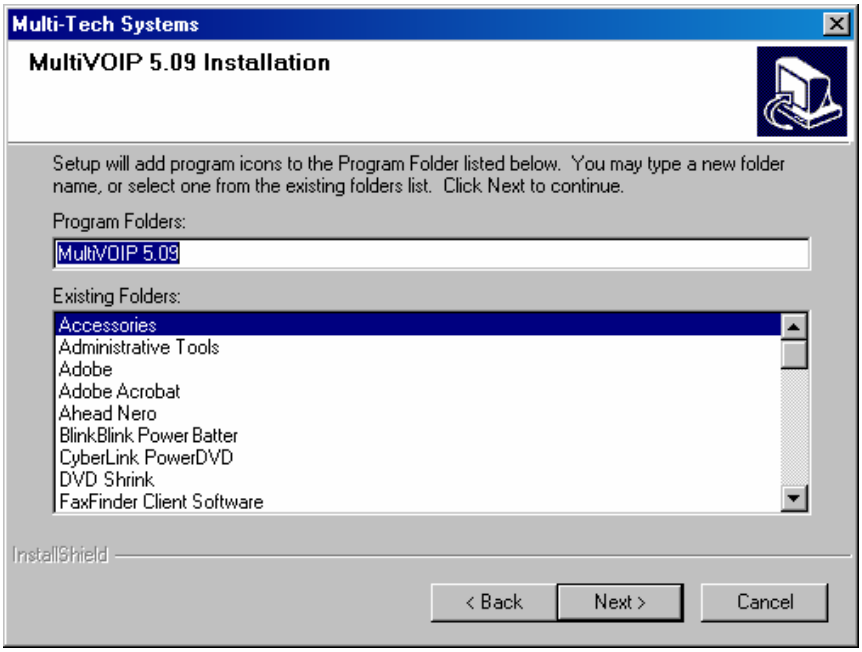
Press **Enter** or click **Next** to continue.

5. Follow the on-screen instructions to install your MultiVOIP software. The first screen asks you to choose the folder location of the files of the MultiVOIP software.


Multi-Tech Systems
MultiVOIP 5.09 Installation
Setup will install MultiVOIP in the following folder.
To install to this folder, Click Next. To install to another folder, Click Browse and select another folder.
Destination Folder
C:\Program Files\Multi-Tech Systems\MultiVOIP 5.09
Browse...
InstallShield
< Back
Next >
Cancel

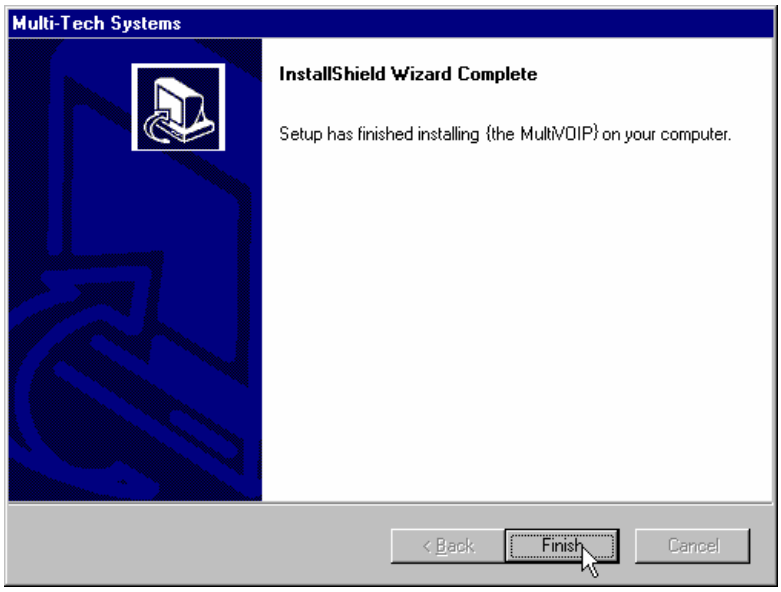
Choose a location and click **Next.**

6. At the next screen, you must select a program folder location for the MultiVOIP software program icon.



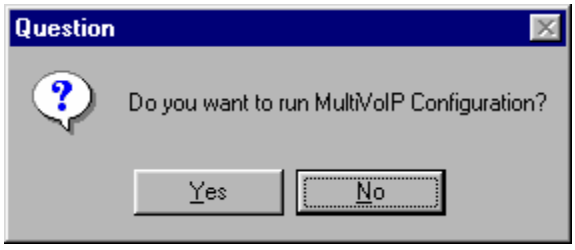
Click **Next**. Transient progress screens will appear while files are being copied.

7. A completion screen will appear.



Click **Finish**.

8. When setup of the MultiVOIP software is complete, you will be prompted to run the MultiVOIP software to configure the VOIP.

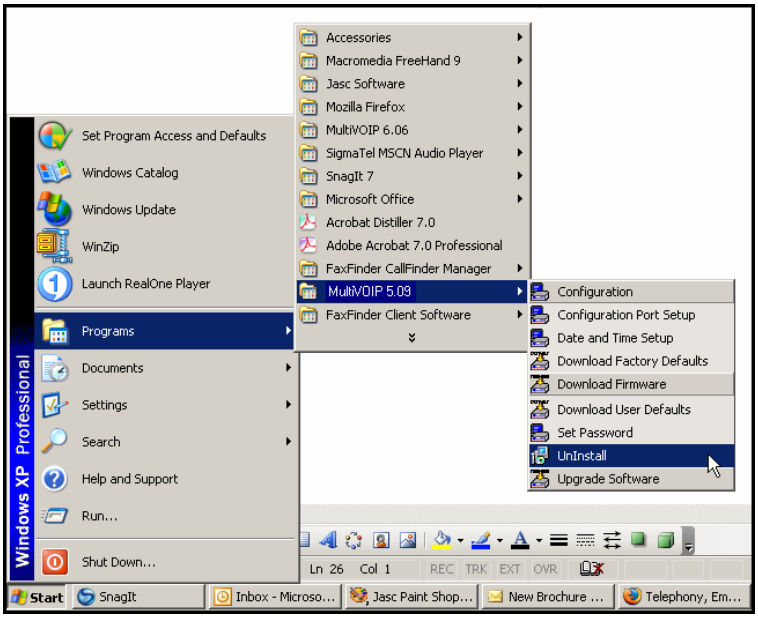


Software installation is complete at this point. You may proceed with Technical Configuration now, or not, at your convenience.

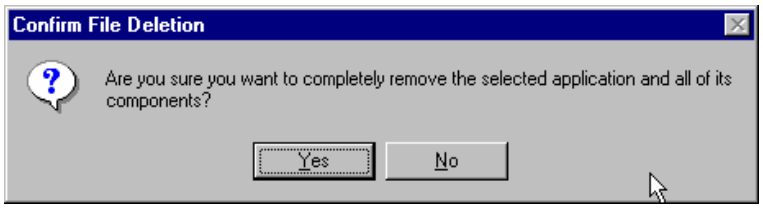
Technical Configuration instructions are in the next chapter of this manual.

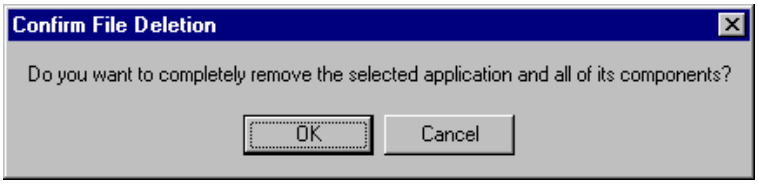
Un-Installing the MultiVOIP Configuration Software

1. To un-install the MultiVOIP configuration software, go to **Start** | **Programs** and locate the entry for the MultiVOIP program. Select **Uninstall**.

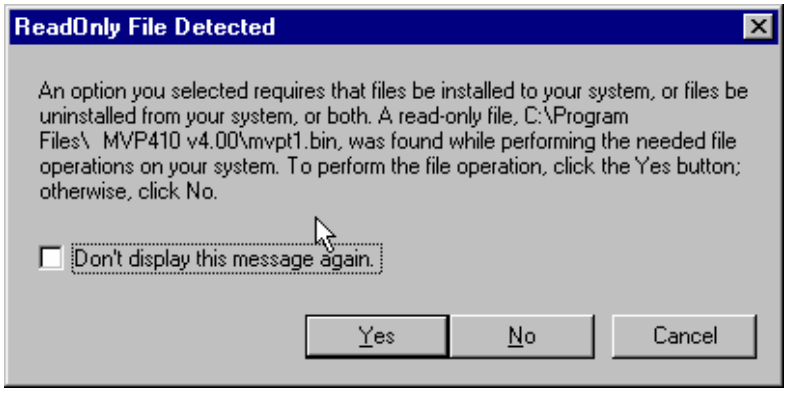


2. Two confirmation screens will appear. Click **Yes** and **OK** when you are certain you want to continue with the uninstallation process.

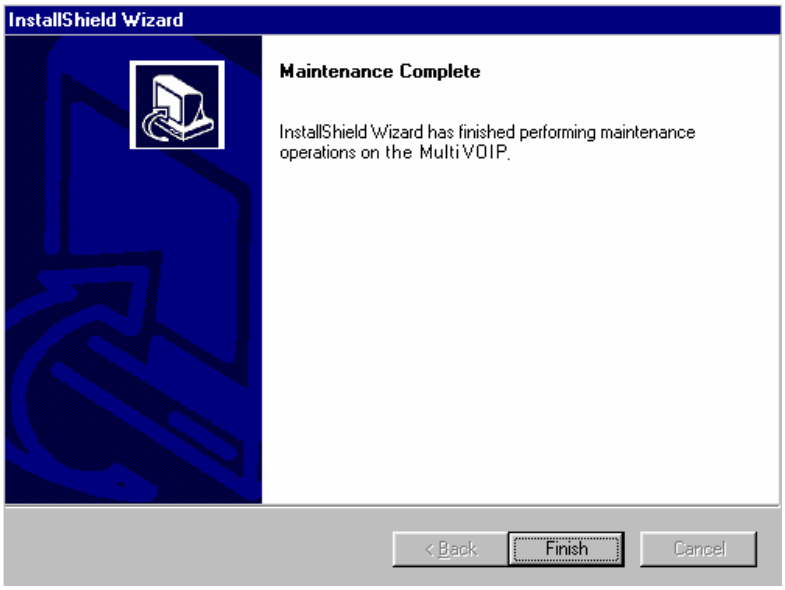

Confirm File Deletion
Are you sure you want to completely remove the selected application and all of its components?
Yes
No



3. A special warning message similar to that shown below may appear concerning the MultiVOIP software's ".bin" file. Click **Yes**.


ReadOnly File Detected
An option you selected requires that files be installed to your system, or files be uninstalled from your system, or both. A read-only file, C:\Program Files\ MVP410 v4.00\mvpt1.bin, was found while performing the needed file operations on your system. To perform the file operation, click the Yes button; otherwise, click No.
Don't display this message again.
Yes
No
Cancel

4. A completion screen will appear.



Click **Finish**.

Chapter 5: Technical Configuration

Configuring the MultiVOIP

There are two ways in which the MultiVOIP must be configured before operation: technical configuration and phonebook configuration.

Technical Configuration. First, the MultiVOIP must be configured to operate with technical parameter settings that will match the equipment with which it interfaces. There are eight types of technical parameters that must be set.

These technical parameters pertain to
(1) its operation in an IP network,
(2) its operation with telephony equipment,
(3) its transmission of voice and fax messages,
(4) its interaction with SNMP (Simple Network Management Protocol) network management software (NOTE: SNMP is not currently supported for BRI MultiVOIPs),
(5) certain telephony attributes that are common to particular nations or regions,
(6) its operation with a mail server on the same IP network (per SMTP parameters) such that log reports about VoIP telephone call traffic can be sent to the administrator by email,
(7) implementing some common premium telephony features (Call Transfer, Call Hold, Call Waiting, Call ID – "Supplementary Services"), and
(8) selecting the method by which log reports will be made accessible.

The process of specifying values for the various parameters in these seven categories is what we call "technical configuration" and it is described in this chapter.

Phonebook Configuration. The second type of configuration that is required for the MultiVOIP pertains to the phone number dialing sequences that it will receive and transmit when handling calls. Dialing patterns will be affected by both the PBX/telephony equipment and the other VOIP devices that the MultiVOIP unit interacts with. We call this "Phonebook Configuration," and, for analog MultiVOIP units, it is described in Chapter 6. The *Quick Start Guide* presents additional information on phonebook setup.

Local/Remote Configuration. The MultiVOIP is typically configured locally at first (to establish an IP address for the MultiVOIP unit). However, you can configure the MultiVOIP remotely from the start by using the web browser GUI and accessing the MultiVOIP unit at its default IP address, which is 192.168.3.143. (Remember, however, that

192.168.3.143 is the default IP address set at the factory for all MultiVOIP units. Once you have established contact, you will need to set up a different IP address that better fits your system.) Changes to this initial configuration can be done either locally or remotely.

Local configuration is done through a connection between the "Command" port of the MultiVOIP and the COM port of the computer; the MultiVOIP configuration program is used.

Remote configuration is done through a connection between the MultiVOIP's Ethernet (network) port and a computer connected to the same network. The computer could be miles or continents away from the MultiVOIP itself. The MultiVOIP web browser interface program is the only remote configuration program that is supported for the BRI MultiVOIP units. There is a GUI screen for SNMP, which relates to the MultiVoipManager program. However, neither SNMP nor MultiVoipManager are supported for the BRI MultiVOIP units.

Web Browser Interface. The MultiVOIP web browser GUI gives access to the same commands and configuration parameters as are available in the MultiVOIP Windows GUI except for logging functions. When using the web browser GUI, logging can be done by email (the SMTP option).

Functional Equivalence of Interfaces. The MultiVOIP Windows configuration program is generally used to do the initial configuration (that is, setting an IP address for the MultiVOIP unit) so that the VOIP unit can communicate with the the web browser GUI. Management of the VOIP after that point can be done from either of these two programs since they both offer essentially the same functionality. Functionally, the web browser GUI can replace the MultiVOIP configuration program after the initial configuration is complete (with minor exceptions, as noted, especially logging).

WARNING: Do not attempt to interface the MultiVOIP unit with two control programs simultaneously (that is, by accessing the MultiVOIP Windows configuration program via the Command Port and the web browser interface via the Ethernet Port). The results of using two programs to control a single VOIP simultaneously would be unpredictable.

Local Configuration

This manual primarily describes local configuration with the Windows GUI. After IP addresses have been set locally using the Windows GUI, most aspects of configuration (logging functions are an exception) can be handled through the web browser GUI, as well (see the *Operation and Maintenance* chapter of this manual). In most aspects of configuration, the Windows GUI and web-browser GUI differ only graphically, not functionally.

Pre-Requisites

To complete the configuration of the MultiVOIP unit, you ***must*** know several things about the overall system.

Before configuring your MultiVOIP Gateway unit, you must know the values for several IP and telephone parameters that describe the IP network system and telephony system (PBX or telco central office equipment) with which the digital MultiVOIP will interact. If you plan to receive log reports on phone traffic by email (SMTP), you must arrange to have an email address assigned to the VOIP unit on the email server on your IP network. A summary of this configuration information appears on page 46 ("Config Info CheckList").

IP Parameters

The following parameters must be known about the network (LAN, WAN, Internet, etc.) to which the MultiVOIP will connect:

➳	*Ask your computer network administrator.*	*Info needed to operate:* **all MultiVOIP models.**
	IP Network Parameters: Record for each VOIP Site in System	
	• IP Address	
	• IP Mask	
	• Gateway	
	• Domain Name Server (DNS) Info	
	• If SIP protocol is used, determine whether or not 802.1p Packet Prioritization will be used.	

Write down the values for these IP parameters. You will need to enter these values in the "IP Parameters" screen in the Configuration section of the MultiVOIP software. You must have this IP information about *every* VOIP in the system.

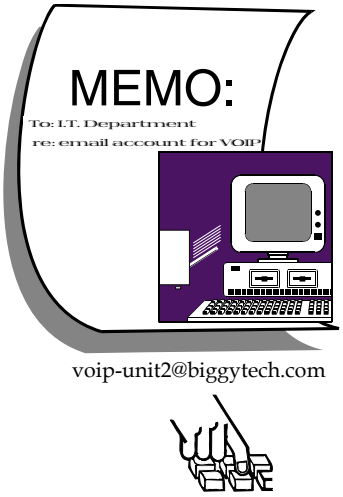
ISDN-BRI Telephony Parameters

The following parameters must be known about the PBX or telco central office equipment to which the analog MultiVOIP will connect:

➳	**ISDN-BRI Phone Parameters** *Ask phone company or telecom manager.*	
	ISDN-BRI Telephony Interface Parameters: Record them for this VOIP Site	
	• In which country is this voip installed?	
	• Which operator (switch type) is used?	
	• What type of line coding use required, A-law or u-law?	
	• Determine which BRI ports will be network side and which BRI ports will be terminal side.	

Write down the values for these ISDN/BRI telephony parameters. You will need to enter these values in the "ISDN Parameters" screen (ISDN-BRI) in the Configuration section of the MultiVOIP software.

SMTP Parameters (for email call log reporting)

required if log reports of VOIP call traffic are to be sent by email	**Optional**
SMTP Parameters Preparation Task: Ask Mail Server administrator to set up email account (with password) for the MultiVOIP unit itself. Be sure to give a unique identifier to each individual MultiVOIP unit. . Get the IP address of the mail server computer, as well.	

Config Info CheckList

Type of Config Info Gathered	MultiVOIP Configuration screen on which to enter Config Info	Info Obtained √	Info Entered
IP info for voip unit ● IP address ● Gateway ● DNS IP (if used) ● 802.1p Prioritization (if used)	Ethernet/IP Parameters		
ISDN Layer 1 Interface (Choices: Network, Terminal)	ISDN BRI Parameters		
Clock Master Status (only a channel designated as Terminal can be used as the Clock Master)	ISDN BRI Parameters		
Switch Info ● Country ● Operator ● TEI Assignment ● A-Law or Mu-Law?	ISDN BRI Parameters		
Calling Party Number Type	ISDN BRI Parameters		
Called Party Number Type & Plan	ISDN BRI Parameters		
Country Code	Regional Parameters		
Email address for voip (optional)	SMTP Parameters		
Reminder: *Be sure to* **Save Setup** *after entering configuration values.*			

Local Configuration Procedure (Summary)

After the MultiVOIP configuration software has been installed in the 'Command' PC (which is connected to the MultiVOIP unit), several steps must be taken to configure the MultiVOIP to function in its specific setting. Although the summary below includes all of these steps, some are optional.

1. Check Power and Cabling.

2. Start MultiVOIP Configuration Program.

3. Confirm Connection.

4. Solve Common Connection Problems.

   A. Fixing a COM Port Problem.

   B. Fixing a Cabling Problem.

5. Familiarize yourself with configuration parameter screens and how to access them.

6. Set Ethernet/IP Parameters.

7. Enable web browser GUI (optional).

8. Set Voice/Fax Parameters.

9. Set Call Signaling parameters. The choice of H.323, SIP, or SPP is made in the Outbound Phonebook, but details are configured in the Call Signaling Parameters screen.

10. Set ISDN/BRI Parameters.

11. Set SNMP Parameters (not supported for BRI MultiVOIPs).

12. Set Regional Parameters (Phone Signaling Tones & Cadences).

13. Set Custom Tones and Cadences (optional).

14. Set SMTP Parameters (applicable if Log Reports are via Email).

15. Set RADIUS parameters. Optional. Used only if system interfaces with RADIUS server for billing or other accounting functions.

16. Set Log Reporting Method (GUI, locally in MultiVOIP Configuration program; SNMP, remotely in MultiVoipManager program; or SMTP, via email).

17. Set NAT Traversal (STUN) parameters. Optional. Applicable only under SIP Call Signaling when the UDP transport protocol is used.

18. Set Supplementary Services Parameters. The Supplementary Services screen allows voip deployment of features that are normally found in PBX or PSTN systems (e.g., call transfer and call waiting).

19. View System Info screen and set the updating interval (optional).

20. Set Baud Rate (of COM port connection to 'Command' PC).

21. Save the MultiVOIP configuration.

22. Create a User Default Configuration (optional).

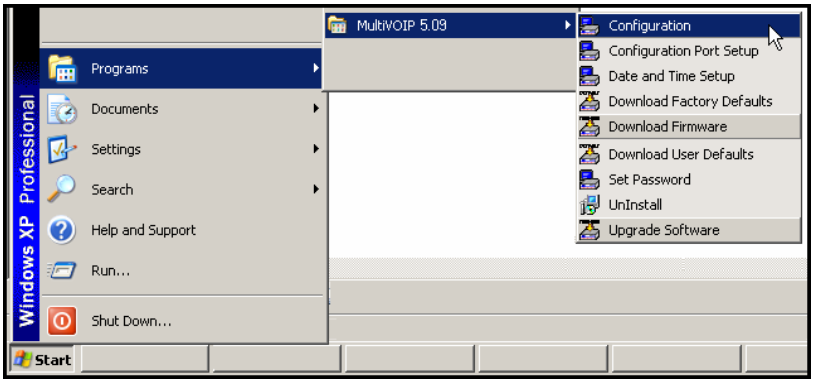
When technical configuration is complete, you will need to configure the MultiVOIP's inbound and outbound phonebooks. This manual has separate chapters describing ***T1 Phonebook Configuration*** for North-American-influenced telephony settings and ***E1 Phonebook Configuration*** for Euro-influenced telephony settings.

Local Configuration Procedure (Detailed)

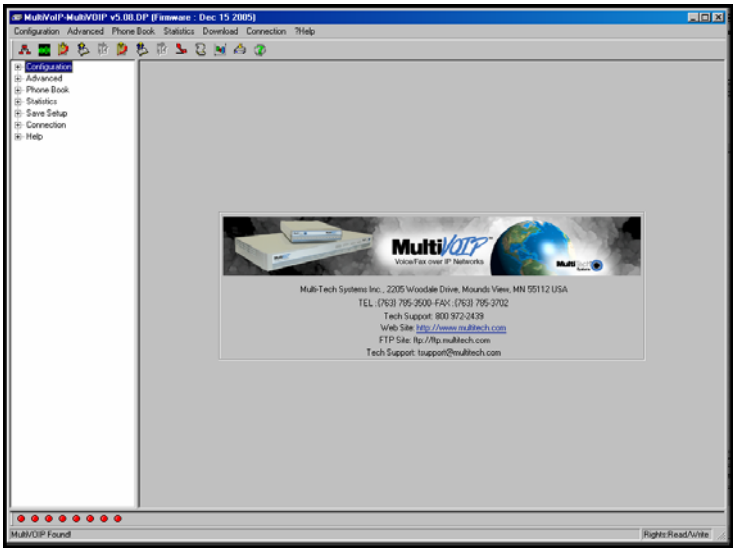
You can begin the configuration process as a continuation of the MultiVOIP software installation. You can establish your configuration or modify it at any time by launching the MultiVOIP program from the Windows **Start** menu.

1. **Check Power and Cabling**. Be sure the MultiVOIP is turned on and connected to the computer via the MultiVOIP's Command Port (DB9 connector at computer's COM port – or at USB2.0 port using serial-to-USB2.0 adaptor if no serial port is available -- and RJ45 connector at MultiVOIP).

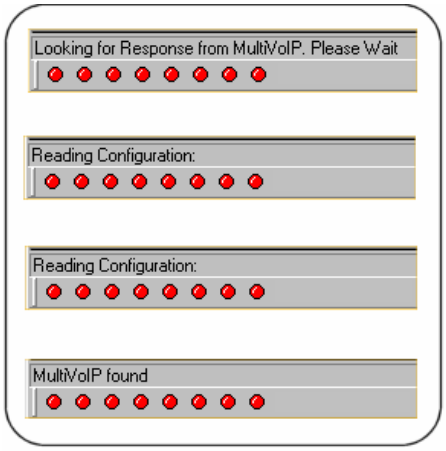
2. **Start MultiVOIP Configuration Program**. Launch the MultiVOIP program from the Windows **Start** menu (from the folder location determined during installation).



3. **Confirm Connection**. If the MultiVOIP is set for an available COM port and is correctly cabled to the PC, the MultiVOIP main screen will appear. (If the main screen appears *grayed out* and seems inaccessible, go to step 4.)



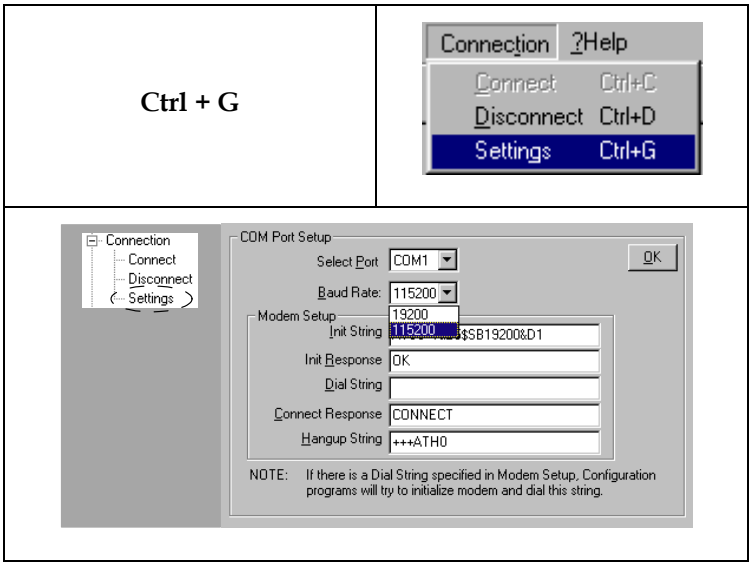
In the lower left corner of the screen, the connection status of the MultiVOIP will be displayed. The messages in the lower left corner will change as detection occurs. The message "MultiVOIP Found" confirms that the MultiVOIP is in contact with the MultiVOIP configuration program. Skip to step 5.



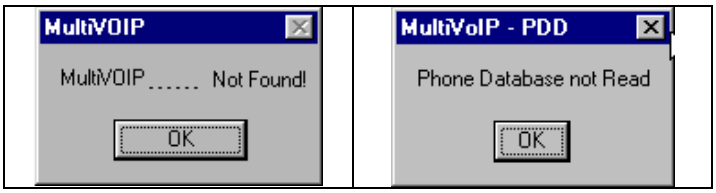
4. **Solving Common Connection Problems**.

A. Fixing a COM Port Problem. If the MultiVOIP main screen appears but is grayed out and seems inaccessible, the COM port that was specified for its communication with the PC is unavailable and must be changed. An error message will appear.

To change the COM port setting, use the **COM Port Setup** dialog box, which is accessible via the keyboard shortcut **Ctrl + G** or by going to the **Connection** pull-down menu and choosing "Settings." In the "Select Port" field, select a COM port that is available on the PC. (If no COM ports are currently available, re-allocate COM port resources in the computer's MS Windows operating system to make one available.)



4B. Fixing a Cabling Problem. If the MultiVOIP cannot be located by the computer, two error messages will appear (saying "Multi-VOIP Not Found" and "Phone Database Not Read").

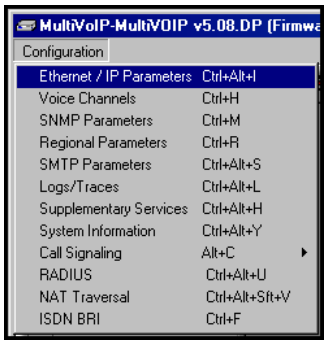
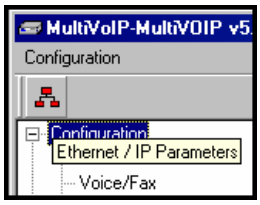
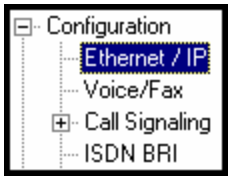


In this case, the MultiVOIP is simply disconnected from the network. For instructions on MultiVOIP cable connections, see the Cabling section of Chapter 3.

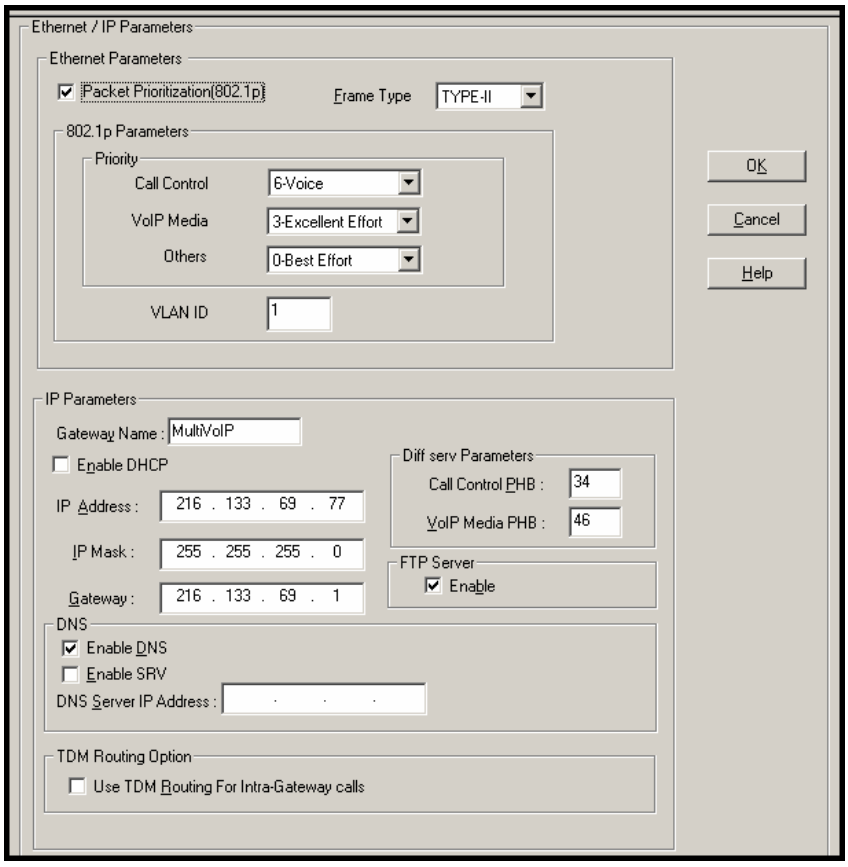
5. **Configuration Parameter Groups: Getting Familiar, Learning About Access**. The first part of configuration concerns IP parameters, Voice/FAX parameters, Telephony Interface parameters, SNMP parameters, Regional parameters, SMTP parameters, Supplementary Services parameters, Logs, and System Information. In the MultiVOIP software, these seven types of parameters are grouped together under "Configuration" and each has its own dialog box for entering values.

Generally, you can reach the dialog box for these parameter groups in one of four ways: pulldown menu, toolbar icon, keyboard shortcut, or sidebar.

6. **Set Ethernet/IP Parameters.** This dialog box can be reached by pulldown menu, toolbar icon, keyboard shortcut, or sidebar.

<table>
<tr><th colspan="2">Accessing “Ethernet/IP Parameters”</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td> 
</td><td> 
</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>Ctrl + Alt + I</td><td> 
</td></tr>
</table>

In each field, enter the values that fit your particular network.



The **Ethernet/IP Parameters** fields are described in the tables and text passages below. Note that both DiffServ parameters (Call Control PHB and VoIP Media PHB) must be set to zero if you enable Packet Prioritization (802.1p). Nonzero DiffServ values negate the prioritization scheme.

Ethernet/IP Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Ethernet Parameters		
Packet Prioritization (802.1p)	Y/N	Select to activate prioritization under 802.1p protocol (described below). .
Frame Type	Type II, SNAP	Must be set to match network's frame type. Default is Type II.
802.1p	A draft standard of the IEEE about data traffic prioritization on Ethernet networks. The 802.1p draft is an extension of the 802.1D bridging standard. 802.1D determines how prioritization will operate within a MAC-layer bridge for any kind of media. The 802.1Q draft for virtual local-area-networks (VLANs) addresses the issue of prioritization for Ethernet networks in particular. 802.1p enacts this Quality-of-Service feature using 3 bits. This 3-bit code allows data switches to reorder packets based on priority level. The descriptors for the 8 priority levels are given below. **802.1p PRIORITY LEVELS** *LOWEST PRIORITY* 1 – **Background**: Bulk transfers and other activities permitted on the network, but should not affect the use of network by other users and applications. 2 – **Spare**: An unused (spare) value of the user priority. 0 – **Best Effort** (default): Normal priority for ordinary LAN traffic. 3 – **Excellent Effort**: The best effort type of service that an information services organization would deliver to its most important customers.	

Ethernet/IP Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Ethernet Parameters		
802.1p (continued)	4 – **Controlled Load**: Important business applications subject to some form of "Admission Control", such as preplanning of Network requirement, characterized by bandwidth reservation per flow. 5 – **Video**: Traffic characterized by delay < 100 ms. 6 – **Voice**: Traffic characterized by delay < 10 ms. 7 - **Network Control**: Traffic urgently needed to maintain and support network infrastructure. *HIGHEST PRIORITY*	
Call Control Priority	0-7, where 0 is lowest priority	Sets the priority for signaling packets.
VoIP Media Priority	0-7, where 0 is lowest priority	Sets the priority for media packets.
Others (Priorities)	0-7, where 0 is lowest priority	Sets the priority for SMTP, DNS, DHCP, and other packet types.
VLAN ID	1 - 4094	The 802.1Q IEEE standard allows virtual LANs to be defined within a network. This field identifies each virtual LAN by number.

Ethernet/IP Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
IP Parameter fields		
Gateway Name	alphanumeric	Descriptor of current voip unit to distinguish it from other units in system.
Enable DHCP	Y/N disabled by default	Dynamic Host Configuration Protocol is a method for assigning IP address and other IP parameters to computers on the IP network in a single message with great flexibility. IP addresses can be static or temporary depending on the needs of the computer.
IP Address	4-places, 0-255	The unique LAN IP address assigned to the MultiVOIP.
IP Mask	4-places, 0-255	Subnetwork address that allows for sharing of IP addresses within a LAN.
Gateway	4-places, 0-255.	The IP address of the device that connects your MultiVOIP to the Internet.

Ethernet/IP Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
DiffServ Parameter fields	DiffServ PHB (Per Hop Behavior) values pertain to a differential prioritizing system for IP packets as handled by DiffServ-compatible routers. There are 64 values, each with an elaborate technical description. These descriptions are found in TCP/IP standards RFC2474, RFC2597, and, for present purposes, in RFC3246, which describes the value 34 (34 decimal; 22 hex) for Assured Forwarding behavior (default for Call Control PHB) and the value 46 (46 decimal; 2E hexadecimal) for Expedited Forwarding behavior (default for Voip Media PHB). Before using values other than these default values of 34 and 46, consult these standards documents and/or a qualified IP telecommunications engineer. To disable DiffServ, configure both fields to 0 decimal. The next page explains DiffServ in the context of the IP datagram.	
Call Control PHB	0 – 63 default = 34 .	Value is used to prioritize call setup IP packets.
Voip Media PHB	0 – 63 default = 46 *n*	Value is used to prioritize the RTP/RTCP audio IP packets.

The IP Datagram with Header, Its Type-of-Service field, & DiffServ

bits =>

0	4	8	16	19	24	31
VERS	HLEN	TYPE OF SERVICE	TOTAL LENGTH			
IDENTIFICATION			FLAGS	FRAGMENT OFFSET		
TIME TO LIVE	PROTOCOL		HEADER CHECKSUM			
SOURCE IP ADDRESS						
DESTINATION IP ADDRESS						
IP OPTIONS (if any)					PADDING … ***end of header***	
DATA						
…						

The TOS field consists of eight bits, of which only the first six are used. These six bits are called the "Differentiated Service Codepoint" or DSCP bits.

The Type of Service or "TOS" field

0	1	2	3	4	5	6	7
PRECEDENCE			D	T	R	*unused*	

three precedence have eight values, 0-7, ranging from "normal" precedence (value of 0) to "network control" (value of 7). When set, the *D* bit requests low delay, the *T* bit requests high throughput, and the *R* bit requests high reliability.

Routers that support DiffServ can examine the six DSCP bits and prioritize the packet based on the DSCP value. The DiffServ Parameters fields in the MultiVOIP IP Parameters screen allow you to configure the DSCP bits to values supported by the router. Specifically, the Voip Media PHB field relates to the prioritizing of audio packets (RTP and RTCP packets) and the Call Control PHB field relates to the prioritzing of non-audio packets (packets concerning call set-up and tear-down, gatekeeper registration, etc.).

The MultiVOIP Call Control PHB parameter defaults to 34 decimal (22 hex; 100010 binary – consider vis-à-vis TOS field above) for Assured Forwarding behavior. The MultiVOIP Voip Media PHB parameter defaults to the value 46 decimal (2E hex; 101110 binary – consider vis-à-vis TOS field above). To disable DiffServ, configure both fields to 0 decimal.

Ethernet/IP Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
FTP Parameter fields		
FTP Server Enable	Y/N Default = disabled See "FTP Server File Transfers" in *Operation & Maintenance* chapter.	MultiVOIP unit has an FTP Server function so that firmware and other important operating software files can be transferred to the voip via the network.
DNS Parameter fields		
Enable DNS	Y/N Default = disabled	Enables Domain Name Space/System function where computer names are resolved using a worldwide distributed database.
Enable SRV	Y/N	Enables 'service record' function. Service record is a category of data in the Internet Domain Name System specifying information on available servers for a specific protocol and domain, as defined in RFC 2782. Newer internet protocols like SIP, STUN, H.323, POP3, and XMPP may require SRV support from clients. Client implementations of older protocols, like LDAP and SMTP, may have been enhanced in some settings to support SRV.
DNS Server IP Address	4-places, 0-255.	IP address of specific DNS server to be used to resolve Internet computer names.

About Service Records

An SRV record holds the following information:

- **Service**: the symbolic name of the desired service.
- **Protocol**: this is usually either TCP or UDP.
- **Domain name**: the domain for which this record is valid.
- **TTL**: standard DNS time to live field.
- **Class**: standard DNS class field (this is always *IN*).
- **Priority**: the priority of the target host.
- **Weight**: A relative weight for records with the same priority.
- **Port**: the TCP or UDP port on which the service is to be found.
- **Target**: the hostname of the machine providing the service.

An example SRV record might look like this:

```
_sip._tcp.example.com 86400 IN SRV 0 5 5060 sipserver.example.com.
```

This expression denotes a server named sipserver.example.com. This server listens on TCP port 5060 for SIP protocol connections. The priority given here is 0, and the weight is 5.

7. **Set up the Web Browser GUI (Optional)**. After an IP address for the MultiVOIP unit has been established, you can choose to do any further configuration of the unit (a) by using the MultiVOIP web browser GUI, or (b) by continuing to use the MultiVOIP Windows GUI. If you want to do configuration work using the web browser GUI, you must first set it up. To do so, follow the steps below.

A. Set IP address of MultiVOIP unit using the MultiVOIP Configuration program (the Windows GUI).

B. Save Setup in Windows GUI.

C. Close Windows GUI.

D. Install Java program from MultiVOIP product CD (on first use only).

E. Open web browser.

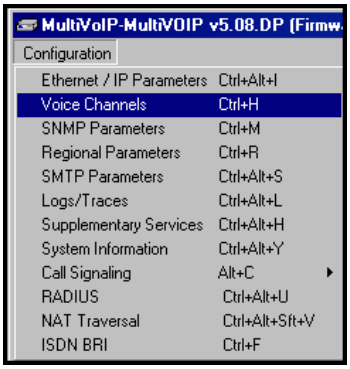
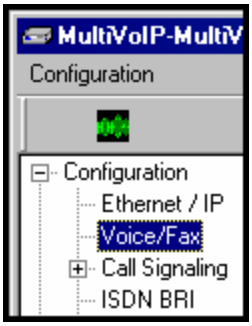
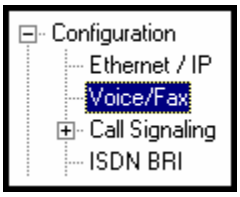
F. Browse to IP address of MultiVOIP unit.

G. If username and password have been established, enter them when when prompted.

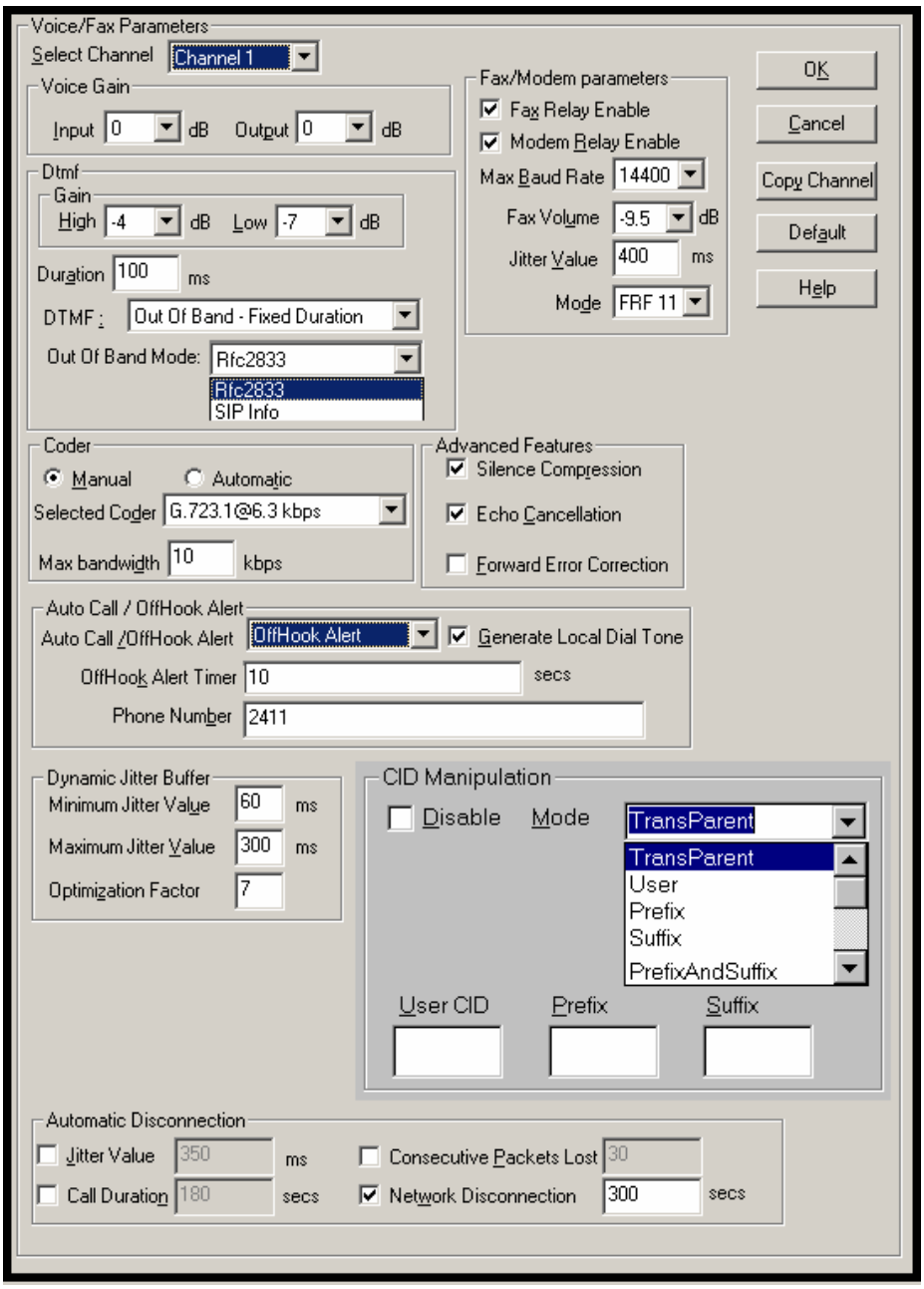
H. Use web browser GUI to configure or operate MultiVOIP unit. The configuration screens in the web browser GUI will have the same content as their counterparts in the Windows GUI; only the graphic presentation will be different.

For more details on enabling the MultiVOIP web GUI, see the "Web Browser Interface" section of the *Operation & Maintenance* chapter of this manual.

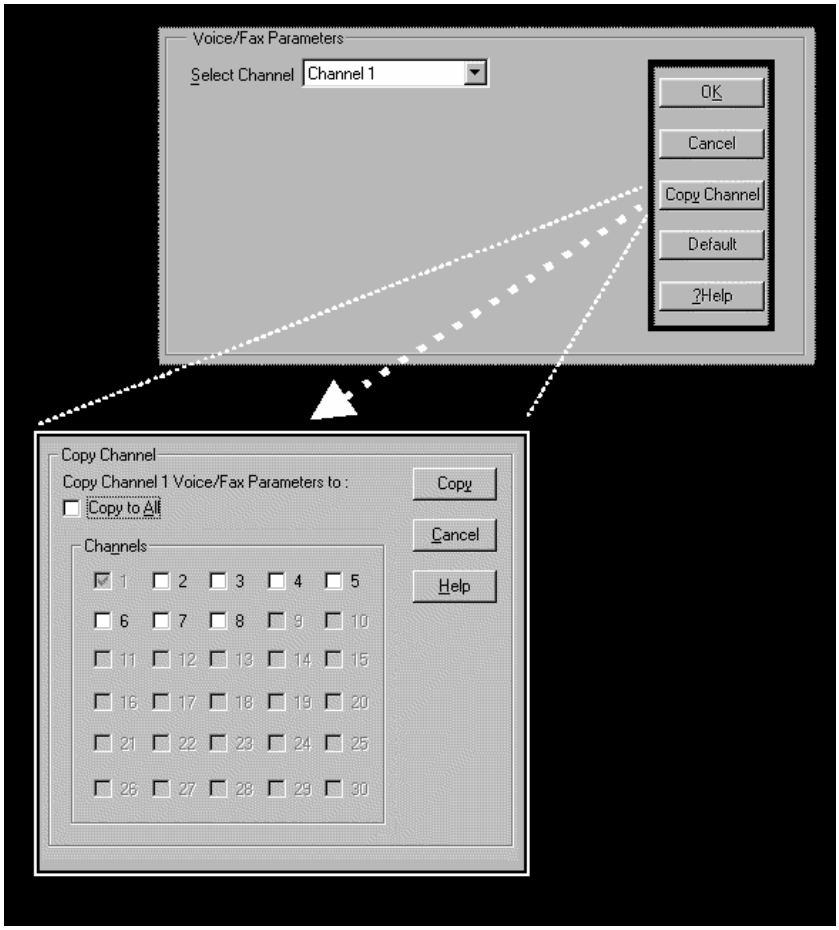
8. **Set Voice/FAX Parameters.** This dialog box can be reached by pulldown menu, toolbar icon, keyboard shortcut, or sidebar.

<table>
<tr><th colspan="2">Accessing “Voice/FAX Parameters”</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td> 
</td><td> 
</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>Ctrl + H</td><td> 
</td></tr>
</table>

In each field, enter the values that fit your particular network.



Note that Voice/FAX parameters are applied on a channel-by-channel basis. However, once you have established a set of Voice/FAX parameters for a particular channel, you can apply this entire set of Voice/FAX parameters to another channel by using the **Copy Channel** button and its dialog box. To copy a set of Voice/FAX parameters to all channels, select "Copy to All" and click **Copy**.



The **Voice/FAX Parameters** fields are described in the tables below.

Voice/Fax Parameter Definitions		
Field Name	**Values**	**Description**
Default	--	When this button is clicked, all Voice/FAX parameters are set to their default values.
Select Channel	1-4 (410ST) 1-8 (810ST)	Channel to be configured is selected here.
Copy Channel	--	Copies the Voice/FAX attributes of one channel to another channel. Attributes can be copied to multiple channels or all channels at once.
Voice Gain	--	Signal amplification (or attenuation) in dB.
Input Gain	+31dB to –31dB	Modifies audio level entering voice channel before it is sent over the network to the remote VOIP. The default & recommended value is **0 dB**.
Output Gain	+31dB to –31dB	Modifies audio level being output to the device attached to the voice channel. The default and recommended value is **0 dB**.
DTMF Parameters		
DTMF Gain	--	The **DTMF Gain** (Dual Tone Multi-Frequency) controls the volume level of the DTMF tones sent out for Touch-Tone dialing.
DTMF Gain, High Tones	+3dB to -31dB & "mute"	Default value: **-4 dB**. Not to be changed except under supervision of MultiTech's Technical Support.
DTMF Gain, Low Tones	+3dB to -31dB & "mute"	Default value: **-7 dB**. Not to be changed except under supervision of MultiTech's Technical Support.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
DTMF Parameters		
Duration (DTMF)	60 – 3000 ms	When **DTMF: Out of Band** is selected, this setting determines how long each DTMF digit 'sounds' or is held. Default = 100 ms. Not supported in 5.02c BRI software.
DTMF In/Out of Band	Out of Band, or Inband	When **DTMF Out of Band** is selected, the MultiVOIP detects DTMF tones at its input and regenerates them at its output. When **DTMF Inband** is selected, the DTMF digits are passed through the MultiVOIP unit as they are received. In 502c BRI software, "DTMF Out of Band" can be checked or unchecked.
Out of Band Mode	RFC 2833, SIP Info	**RFC2833 method**. Uses an RTP mode defined in RFC 2833 to transmit the DTMF digits. **SIP Info method**. Generates dual tone multi frequency (DTMF) tones on the telephony call leg. The SIP INFO message is sent along the signaling path of the call. You must set this parameter per the capabilities of the remote endpoint with which the voip will communicate. The RFC2833 method is the more common of the two methods.
FAX Parameters		
Fax Enable	Y/N	Enables or disables fax capability for a particular channel.
Modem Relay Enable	Y/N	When enabled, modem traffic can be carried on voip system. When disabled, modem traffic will bypass the voip system (Modem Bypass mode).
Max Baud Rate (Fax)	2400, 4800, 7200, 9600, 12000, 14400 bps	Set to match baud rate of fax machine connected to channel (see Fax machine's user manual). Default = 14400 bps.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Valuee**	**Description**
FAX Parameters (cont'd)		
Fax Volume (Default = -9.5 dB)	-18.5 dB to –3.5 dB	Controls output level of fax tones. To be changed only under the direction of Multi-Tech's Technical Support.
Jitter Value (Fax)	Default = 400 ms	Defines the inter-arrival packet deviation (in milliseconds) for the fax transmission. A higher value will increase the delay, allowing a higher percentage of packets to be reassembled. A lower value will decrease the delay allowing fewer packets to be reassembled.
Mode (Fax)	FRF 11; T.38 (T.38 not currently sup-ported)	**FRF11** is frame-relay FAX standard using these coders: G.711, G.728, G.729, G.723.1. **T.38** is an ITU-T standard for storing and forwarding FAXes via email using X.25 packets. It uses T.30 fax standards and includes special provisions to preclude FAX timeouts during IP transmissions.

Voice/Fax Parameter Definitions (cont'd)		
Coder Parameters		
Coder	Manual or Auto-matic	Determines whether selection of coder is manual or automatic. When Automatic is selected, the local and remote voice channels will negotiate the voice coder to be used by selecting the highest bandwidth coder supported by both sides without exceeding the Max Bandwidth setting. G.723, G.729, or G.711 are negotiated.
Selected Coder	**G.711** a/u law 64 kbps; **G.726**, @ 16/24/32 /40 kbps; **G.727**, @ nine bps rates; **G.723.1** @ 5.3 kbps, 6.3 kbps; **G.729**, 8kbps; **Net Coder** @ 6.4, 7.2, 8, 8.8, 9.6 kbps	Select from a range of coders with specific bandwidths. The higher the bps rate, the more bandwidth is used. The channel that you are calling must have the same voice coder selected. Default = G.723.1 @ 6.3 kbps, as required for H.323. Here 64K of digital voice are compressed to 6.3K, allowing several simultaneous conversations over the same bandwidth that would otherwise carry only one. To make selections from the Selected Coder drop-down list, the Manual option must be enabled.
Max bandwidth (coder)	11 – 128 kbps	This drop-down list enables you to select the maximum bandwidth allowed for this channel. The Max Bandwidth drop-down list is enabled only if the Coder is set to Automatic. If coder is to be selected automatically ("Auto" setting), then enter a value for maximum bandwidth.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Advanced Features		
Silence Compression	Y/N	Determines whether silence compression is enabled (checked) for this voice channel. With **Silence Compression** enabled, the MultiVOIP will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth that is being used by the voice channel. Default = on.
Echo Cancellation	Y/N	Determines whether echo cancellation is enabled (checked) for this voice channel. **Echo Cancellation** removes echo and improves sound quality. Default = on.
Forward Error Correction	Y/N	Determines whether forward error correction is enabled (checked) for this voice channel. **Forward Error Correction** enables some of the voice packets that were corrupted or lost to be recovered. FEC adds an additional 50% overhead to the total network bandwidth consumed by the voice channel. Default = Off

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
AutoCall/Offhook Alert Parameters		
Auto Call / Offhook Alert	AutoCall, Offhook Alert	The **AutoCall** option enables the local MultiVOIP to call a remote MultiVOIP without the user having to dial a Phone Directory Database number. As soon as you access the local MultiVOIP voice/fax channel, the MultiVOIP immediately connects to the remote MultiVOIP identified in the **Phone Number** box of this option. If the "Pass Through Enable" field is checked in the Interface Parameters screen, AutoCall must be used. The Offhook Alert option applies only to FXS channels. The **Offhook Alert** option works like this: if a phone goes offhook and yet no number is dialed within a specific period of time (as set in the **Offhook Alert Timer** field), then that phone will automatically dial the Alert phone number for the voip channel. (The Alert phone number must be set in the **Voice/Fax Parameters | Phone Number** field; if the voip system is working without a gatekeeper unit, there must also be a matching phone number entry in the Outbound Phonebook.). One use of this feature would be for emergency use where a user goes off hook but does not dial, possibly indicating a crisis situation. The Offhook Alert feature uses the **Intercept Tone**, as listed in the **Regional Parameters** screen. This tone will be outputted on the phone that was taken off hook but that did not dial. The other end of the connection will hear audio from the "crisis" end as is it would during a normal phone call.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
AutoCall/Offhook Alert Parameters		
Auto Call / Offhook Alert	AutoCall, Offhook Alert	*(continued from previous page)* Both functions apply on a channel-by-channel basis. It would not be appropriate for either of these functions to be applied to a channel that serves in a pool of available channels for general phone traffic. Either function requires an entry in the Outgoing phonebook of the local MultiVOIP and a matched setting in the Inbound Phonebook of the remote voip.
Generate Local Dial Tone	Y/N	*Used for AutoCall only*. If selected, dial tone will be generated locally while the call is being established between gateways. The capability to generate dial tone locally would be particularly useful when there is a lengthy network delay.
Offhook Alert Timer	0 – 3000 seconds	The length of time that must elapse before the offhook alert is triggered and a call is automatically made to the phone number listed in the **Phone Number** field.
Phone Number	--	Phone number used for Auto Call function or Offhook Alert Timer function. This phone number must correspond to an entry in the Outbound Phonebook of the local MultiVOIP and in the Inbound Phonebook of the remote MultiVOIP (unless a gatekeeper unit is used in the voip system).

Voice/Fax Parameter Definitions (cont'd))		
Field Name	**Values**	**Description**
Dynamic Jitter		
Dynamic Jitter Buffer		**Dynamic Jitter** defines a minimum and a maximum jitter value for voice communications. When receiving voice packets from a remote MultiVOIP, varying delays between packets may occur due to network traffic problems. This is called Jitter. To compensate, the MultiVOIP uses a Dynamic Jitter Buffer. The Jitter Buffer enables the MultiVOIP to wait for delayed voice packets by automatically adjusting the length of the Jitter Buffer between configurable minimum and maximum values. An Optimization Factor adjustment controls how quickly the length of the Jitter Buffer is increased when jitter increases on the network. The length of the jitter buffer directly effects the voice delay between MultiVOIP gateways.
Minimum Jitter Value	60 to 400 ms	The minimum dynamic jitter buffer of **60** milliseconds is the minimum delay that would be acceptable over a low jitter network. Default = 150 msec

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Dynamic Jitter		
Maximum Jitter Value	60 to 400 ms	The maximum dynamic jitter buffer of **400** milliseconds is the maximum delay tolerable over a high jitter network. Default = 300 msec
Optimizat-ion Factor	0 to 12	The **Optimization Factor** determines how quickly the length of the Dynamic Jitter Buffer is changed based on actual jitter encountered on the network. Selecting the minimum value of 0 means low voice delay is desired, but increases the possibility of jitter-induced voice quality problems. Selecting the maximum value of 12 means highest voice quality under jitter conditions is desired at the cost of increased voice delay. Default = **7**.

Modem Relay

To place modem traffic onto the voip network (an application called "modem relay"), use Coder G.711 mu-law at 64kbps.

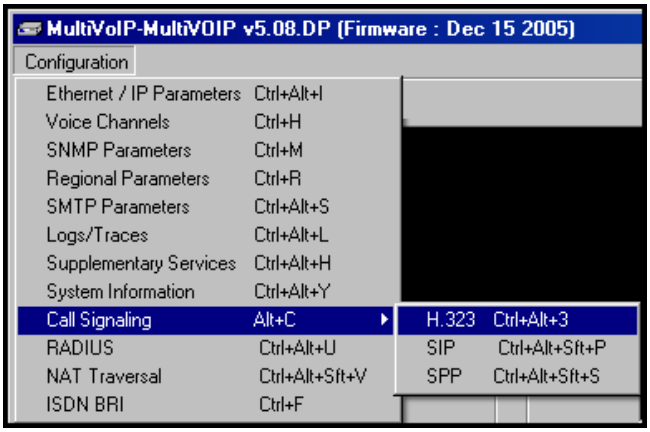
Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
CID Manipulation		
Disable	checked OR unchecked	Caller ID is disabled when checked and enabled when not checked. When CID is disabled, the CID from the BRI line is blocked.
Mode	Transparent, User, Prefix, Suffix, PrefixAnd Suffix	***Transparent***: no changes are made to the caller-ID received from the BRI line. ***User***: The entire caller ID received from the BRI line is replaced with the value listed in the **User** field. ***Prefix***: The value listed in the **Prefix** field is added to the beginning of the caller-ID number that has been received from the BRI line. ***Suffix***: The value listed in the **Suffix** field is added to the end of the caller-ID number received from the BRI line. ***Prefix and Suffix***: The values listed in the **Prefix** and **Suffix** fields are added to the beginning and end (respectively) of the caller-ID number received from the BRI line. Default = Transparent
User CID	string using 0-9, *, and #	The option selected in the **Mode** field determines which of these fields is used with respect to Caller-ID. If "Transparent" is selected, none of these fields is used; instead the Caller-ID from the BRI line is used without alteration. You can use in this field either the values from the "User CID" field, the "Prefix" field, the "Suffix" field, or both the "Prefix" and "Suffix" fields.
Prefix	string using 0-9, *, and #	
Suffix	string using 0-9, *, and #	
PrefixAnd Suffix	two strings using 0-9, *, and #	

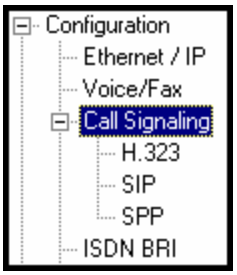
Voice/Fax Parameter Definitions (cont'd))		
Field Name	**Values**	**Description**
Auto Disconnect		
Automatic Disconnect-ion	--	The **Automatic Disconnection** group provides four options which can be used singly or in any combination.
Jitter Value	1-65535 milli-seconds	The **Jitter Value** defines the average inter-arrival packet deviation (in milliseconds) before the call is automatically disconnected. The default is **300** milliseconds. A higher value means voice transmission will be more accepting of jitter. A lower value is less tolerant of jitter. Inactive by default. When active, default = 300 ms. However, value must equal or exceed Dynamic Minimum Jitter Value.
Call Duration	1-65535 seconds	**Call Duration** defines the maximum length of time (in seconds) that a call remains connected before the call is automatically disconnected. Inactive by default. When active, default = **180** sec. This may be too short for most configurations, requiring upward adjustment.
Consecutive Packets Lost	1-65535	**Consecutive Packets Lost** defines the number of consecutive packets that are lost after which the call is automatically disconnected. Inactive by default. When active, default = **30**
Network Discon-nection	1 to 65535 seconds; Default = 30 sec.	Specifies how long to wait before disconnecting the call when IP network connectivity with the remote site has been lost.

9. **Set Call Signaling Parameters**. This dialog box leads to 3 others, one for each of the call-signaling types supported (H.323, SIP, and SPP). These dialog boxes can be reached by pulldown menu, keyboard shortcut, or a sidebar menu.

Accessing "Call Signaling Parameters"

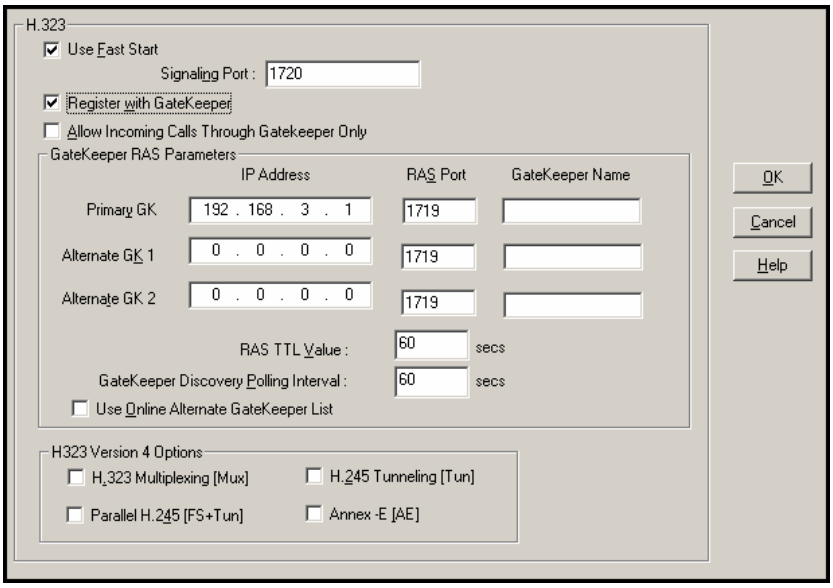
Pulldown



Shortcut	Sidebar
`Alt + C`	

Accessing the Call Signaling Protocols

Protocol	
H.323	Ctrl + Alt + 3
SIP	Ctrl + Alt + Shft + P
SPP	Ctrl + Alt + Shft + S


H.323
Use Fast Start
Signaling Port : 1720
Register with GateKeeper
Allow Incoming Calls Through Gatekeeper Only
GateKeeper RAS Parameters
IP Address
RAS Port
GateKeeper Name
Primary GK 192 . 168 . 3 . 1 1719
Alternate GK 1 0 . 0 . 0 . 0 1719
Alternate GK 2 0 . 0 . 0 . 0 1719
RAS TTL Value : 60 secs
GateKeeper Discovery Polling Interval : 60 secs
Use Online Alternate GateKeeper List
H323 Version 4 Options
H.323 Multiplexing [Mux]
H.245 Tunneling [Tun]
Parallel H.245 [FS+Tun]
Annex -E [AE]
OK
Cancel
Help

The tables below describes all fields in the general **H.323 Call Signaling** screen.

H.323 Call Signaling Parameter Definitions		
Field Name	**Values**	**Description**
Use Fast Start	Y/N	Enables the H.323 Fast Start procedure. May need to be enabled/disabled for compatibility with third-party VOIP gateways.
Signaling Port	port number	Default: 1720 (H.323)
Register with Gatekeeper	Y/N	Check this field to have traffic on current voip gateway controlled by a gatekeeper.
Allow Incoming Calls Through Gatekeeper Only	Y/N	When selected, incoming calls are accepted only if those calls come through the gatekeeper.

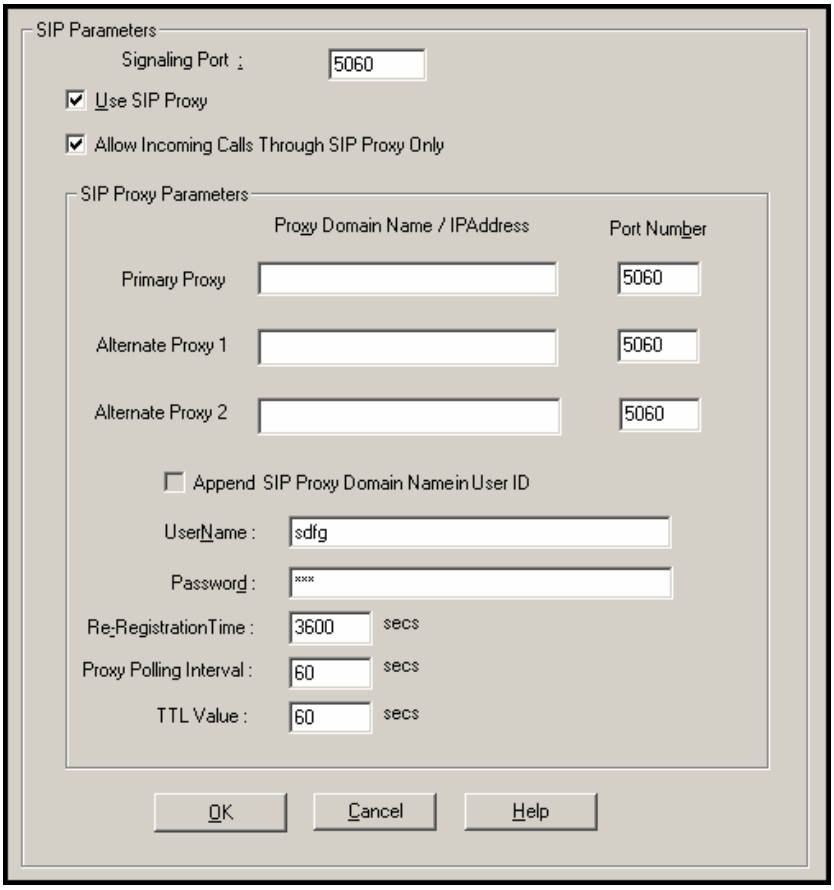
H.323 Call Signaling Parameter Defns (cont'd)		
Field Name	**Values**	**Description**
GateKeeper RAS Parameters		
Primary GK (Gatekeeper)	--	This is the preferred gatekeeper for controlling the traffic of the current voip.
Alternate GK (Gatekeepers) 1 and 2	--	A first and a second alternate gatekeeper can be specified for use by the current voip for situations where the Primary GK is busy or otherwise unavailable.
Gatekeeper / IP Address	n.n.n.n, for n = 0 - 255	IP address of the GateKeeper.
RAS Port	1719	Well-known port number for GateKeepers. Must match port number of GateKeeper, 1719.
Gatekeeper Name	*alpha-numeric string*	Optional. The name of the GateKeeper with which this MultiVOIP is trying to register. A primary gatekeeper and two alternate units are listed.

.

H.323 Call Signaling Parameter Defns (cont'd)		
GateKeeper RAS Parameters		
Field Name	**Values**	**Description**
RAS TTL Value	*in seconds*	The H.323 Gatekeeper "Time to Live" value. As soon as a MultiVOIP gateway registers with a gatekeeper (allowing the gatekeeper to control its call traffic) a countdown timer begins. The RAS TTL Value is the interval of the countdown timer. Before the TTL countdown expires, the MultiVOIP gateway needs to register with the gatekeeper in order to maintain the connection. If the MultiVOIP does not register before the TTL interval expires, the MultiVOIP gateway's registration with the gatekeeper will expire and the gatekeeper will no longer permit call traffic to or from that gateway. Calls in progress will continue to function even if the gateway becomes de-registered.
Gatekeeper Discovery Polling Interval	integer 60 - 300	The interval between the voip gateway's successive attempts to connect to and be governed by a higher level gatekeeper. The Primary GK is the highest level gatekeeper. Alternate GK1 is second; Alternate GK2 is the lowest order gatekeeper.
Use Online Alternate Gatekeeper List (Y/N)	When selected, voip will seek an alternate gatekeeper (when none of the 3 gatekeepers shown on this screen are available) from a list. The list will reside on the Primary gatekeeper or one of the Alternate gatekeepers. The gatekeeper holding the list would download that list onto the voip gateways within the system.	

H.323 Call Signaling Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
H.323 Version 4 Parameters		
H.323 Multiplexing (Mux)	Y/N	Signaling for multiple phone calls can be carried on a single port rather than opening a separate signaling port for each call. This conserves bandwidth resources.
H.245 Tunneling (Tun)	Values: Y/N Description: H.245 messages are encapsulated within the Q.931 call-signaling channel. Among other things, the H.245 messages let the two endpoints tell each other what their technical capabilities are and determine who, during the call, will be the client and who the server. Tunneling is the process of transmitting these H.245 messages through the Q.931 channel. The same TCP/IP socket (or logical port) already being used for the Call Signaling Channel is then also used by the H.245 Control Channel. This encapsulation reduces the number of logical ports (sockets) needed and reduces call setup time.	

H.323 Call Signaling Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
H.323 Version 4 Parameters		
Parallel H.245 (FS + Tun)	Values: Y/N Description: FS (Fast Start or Fast Connect) is a Q.931 feature of H.323v2 to hasten call setup as well as 'pre-opening' the media channel before the CONNECT message is sent. This pre-opening is a requirement for certain billing activities. Under Parallel H.245 FS + Tun, this Fast Connect feature can operate simultaneously with H.245 Tunneling (see description above).	
Annex –E (AE)	Values: Y/N Description: Multiplexed UDP call signaling transport. Annex E is helpful for high-volume voip system endpoints. Gateways with lesser volume can afford to use TCP to establish calls. However, for larger volume endpoints, the call setup times and system resource usage under TCP can become problematic. Annex E allows endpoints to perform call-signaling functions under the UDP protocol, which involves substantially streamlined overhead. (This feature should not be used on the public Internet because of potential problems with security and bandwidth usage.)	

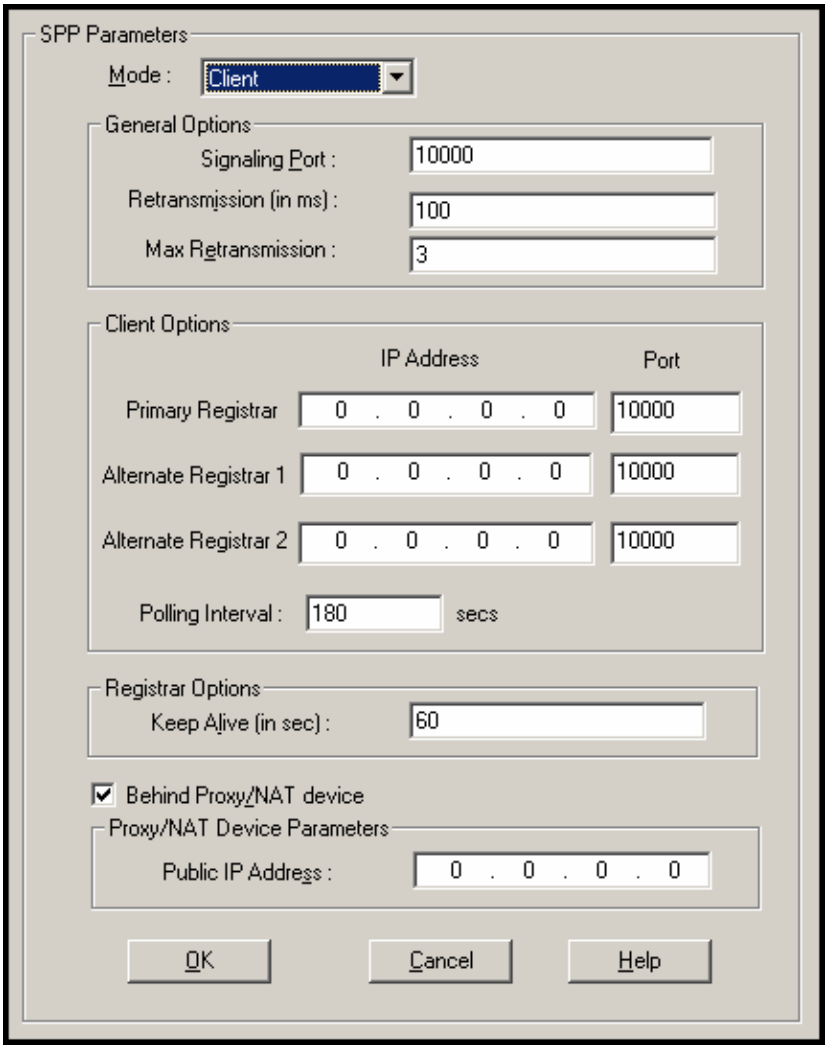


The tables below describes all fields in the general **SIP Call Signaling** screen.

SIP Call Signaling Parameter Definitions		
Field Name	**Values**	**Description**
SIP Proxy Parameters		
Signaling Port		Port number on which the MultiVOIP UserAgent software module will be waiting for any incoming SIP requests.
Use SIP Proxy	Y/N	Allows the MultiVOIP to work in conjunction with a proxy server.

SIP Call Signaling Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
SIP Proxy Parameters		
Allow Incoming Calls Through SIP Proxy Only	Y/N	When selected, incoming calls are accepted only if those calls come through the gatekeeper.
Primary Proxy	--	This is the preferred SIP proxy server for controlling the traffic of the current voip.
Alternate Proxy 1 and 2	--	A first and a second alternate SIP proxy server can be specified for use by the current voip for situations where the Primary proxy server is busy or otherwise unavailable.
Proxy Domain Name / IP Address	n.n.n.n where n=0-255	Network address of the proxy server that the voip is using.
Append SIP Proxy Domain Name in User ID	Y/N	When checked, the domain name of the SIP Proxy serving the MultiVOIP gateway will be included as part of the User ID for that gateway. If unchecked, the SIP Proxy's IP address will be included as part of the User ID instead of the SIP Proxy's domain name.
Port Number		Logical port number for proxy communications.
User Name	Values: alphanumeric Description: Identifier used when proxy server is used in network. If a proxy server is used in a SIP voip network, all clients must enter both a User Name and a Password before being allowed to make a call.	

SIP Call Signaling Parameter Definitions (cont'd)		
Field Name	**Values & Description**	
SIP Proxy Parameters		
Password	Values: alphanumeric Description: Password for proxy server function. See "User Name" description above.	
Re-Registration Time	Values: numeric (in seconds) Description: This is the timeout interval for registration of the MultiVOIP with a SIP proxy server. The time interval begins the moment the MultiVOIP gateway registers with the SIP proxy server and ends at the time specified by the user in the Re-Registration Time field (this field). When/if registration lapses, call traffic routed to/from the MultiVOIP through the SIP proxy server will cease. However, calls in progress will continue to function until they end.	
Proxy Polling Interval	integer 60 - 300	The interval between the voip gateway's successive attempts to connect to and be governed by a higher level SIP proxy server. The Primary Proxy is the highest level gatekeeper. Alternate Proxy 1 is second; Alternate Proxy 2 is the lowest order SIP proxy server.
TTL Value *in seconds*	The SIP proxy "Time to Live" value. As soon as a MultiVOIP gateway registers with a SIP proxy server (allowing the proxy server to control its call traffic) a countdown timer begins. The TTL Value is the interval of the countdown timer. Before the TTL countdown expires, the MultiVOIP gateway needs to register with the gatekeeper in order to maintain the connection. If the MultiVOIP does not register before the TTL interval expires, the MultiVOIP gateway's registration with the proxy server will expire and the proxy server will no longer permit call traffic to or from that gateway. Calls in progress will continue to function even if the gateway becomes de-registered.	


SPP Parameters
Mode : Client
General Options
Signaling Port : 10000
Retransmission (in ms) : 100
Max Retransmission : 3
Client Options
IP Address
Port
Primary Registrar 0 . 0 . 0 . 0 10000
Alternate Registrar 1 0 . 0 . 0 . 0 10000
Alternate Registrar 2 0 . 0 . 0 . 0 10000
Polling Interval : 180 secs
Registrar Options
Keep Alive (in sec) : 60
Behind Proxy/NAT device
Proxy/NAT Device Parameters
Public IP Address : 0 . 0 . 0 . 0
OK
Cancel
Help

The tables below describes all fields in the general **SPP Call Signaling** screen.

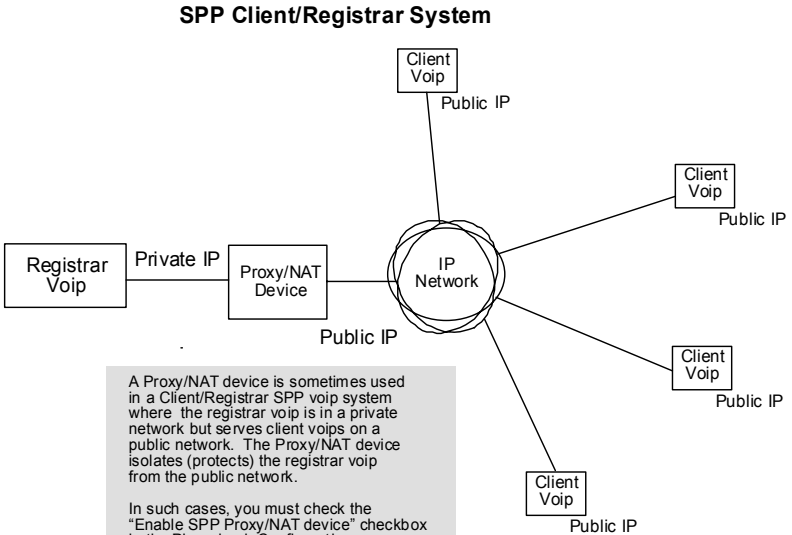
SPP Call Signaling Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Single Port Protocol (SPP)		
Mode	Direct, Client, or Registrar	SPP voip systems can operate in two modes: in the **direct mode**, where all voip gateways have static IP addresses assigned to them; or in the **registrar/client mode**, where one voip gateway serves as registrar and all other gateways, being its clients, point to that registrar. The registrar assigns IP addresses dynamically.
General Options		
Port		The UDP port on which data transmission will occur. Each client voip has its own port. If two client voips are both behind the same firewall, then they must have different ports assigned to them. If there are two clients and each is behind a different firewall, then the clients could have different port numbers or the same port number. (Default port number = 10000.)
Re-trans-mission (in ms)		If packets are lost (as indicated by absence of an acknowledgment) then the endpoint will retransmit the lost packets after this designated time duration has elapsed. (Default value = 2000 milliseconds.)
Max Re-trans-mission		Number of times the voip will re-transmit a lost packet (if no acknowledgment has been received). (Default value = 3)

SPP Call Signaling Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Single Port Protocol (SPP) [continued]		
Client Options		Client Option fields are active only in registrar/client mode and only for client voip units.
Primary Registrar	--	This is the preferred SPP registrar gateway for controlling the traffic of the current voip.
Alternate Registrar 1 and 2	--	A first and a second alternate SPP Registrar gateway can be specified for use by the current voip for situations where the Primary Registrar gateway is busy or otherwise unavailable.
Registrar IP Address	n.n.n.n	This is the IP address of the registrar voip to which this client is assigned. (Default value = 0.0.0.0; effectively, there is no useful default value.)
Registrar Port	10000 or other	This is the port number of the registrar voip to which this client is assigned. (Default port number = 10000.)
Polling Interval	integer 60 - 300	The interval between the voip gateway's successive attempts to connect to and be governed by a higher level SPP registrar gateway. The Primary Registrar is the highest level registrar gateway. Alternate Registrar 1 is second; Alternate Registrar 2 is the lowest order SPP registrar gateway.
Registrar Options		Registrar Option fields are active only in registrar/client mode and only for registrar voip units.
Keep Alive (in sec.)	30 – 300 (seconds)	Time-out duration before a registrar will unregister a client that does not send its "I'm here" signal. Client normally sends its "I'm here" signal every 20 seconds. Timeout default = 60 seconds.

SPP Call Signaling Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Proxy/NAT Device Parameters		
Behind Proxy/NAT device	Y/N	Enables MultiVOIP (running in SPP Registrar mode) to operate 'behind' a proxy/NAT device (NAT = Network Address Translation).
Proxy/NAT Device Parameters – Public IP Address	n.n.n.n where n=0-255	The public IP address of the proxy/NAT device which the MultiVOIP is behind.

An example of a NAT-equipped SPP network is shown below.

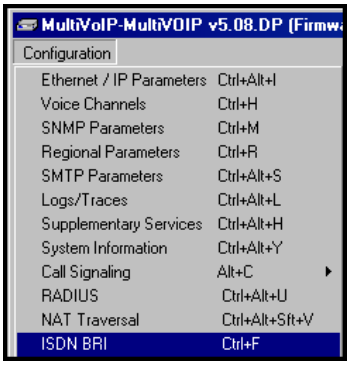
About SPP Proxy/NAT Device Parameters



A Proxy/NAT device is sometimes used in a Client/Registrar SPP voip system where the registrar voip is in a private network but serves client voips on a public network. The Proxy/NAT device isolates (protects) the registrar voip from the public network.

In such cases, you must check the "Enable SPP Proxy/NAT device" checkbox in the Phonebook Configuration screen of the Registrar voip. The private registrar voip can then function with the client voips using the public IP address of the Proxy/NAT device. You must enter this address in the Public IP Address field.

10. **Set ISDN Parameters**. This dialog box can be reached by pulldown menu, keyboard shortcut, or sidebar.

<table>
<tr><th colspan="2">Accessing ISDN (BRI) Parameters</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td> 
</td><td>--</td></tr>
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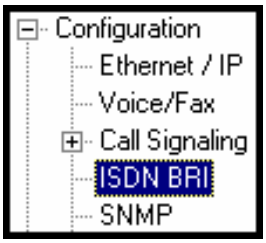
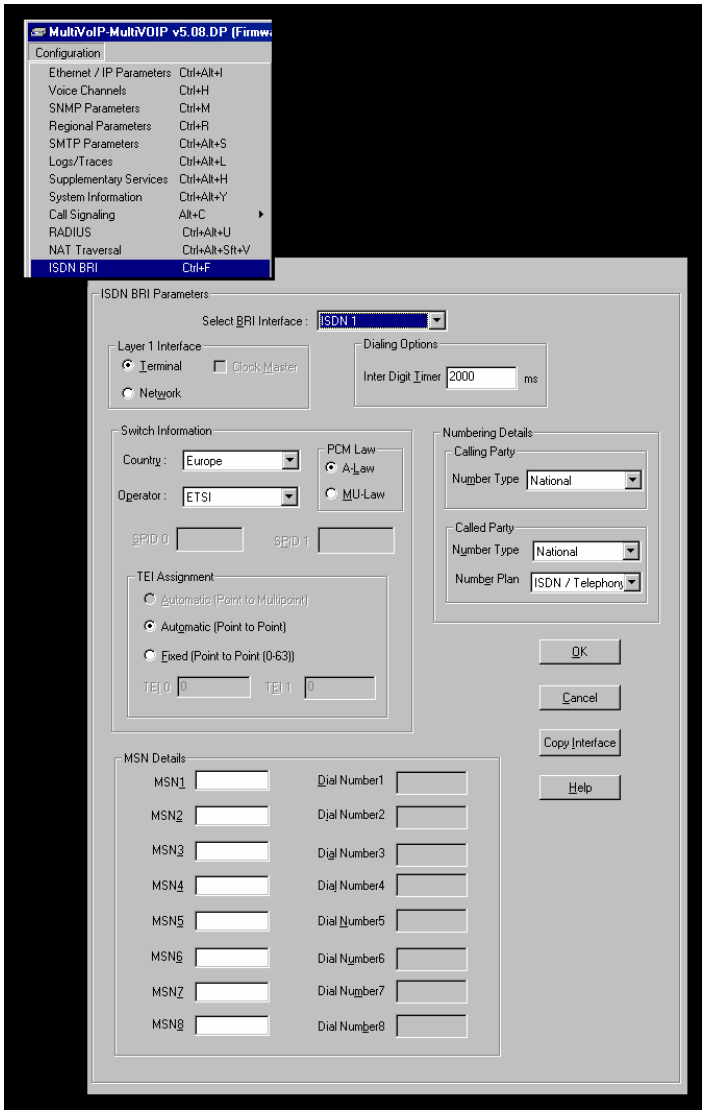
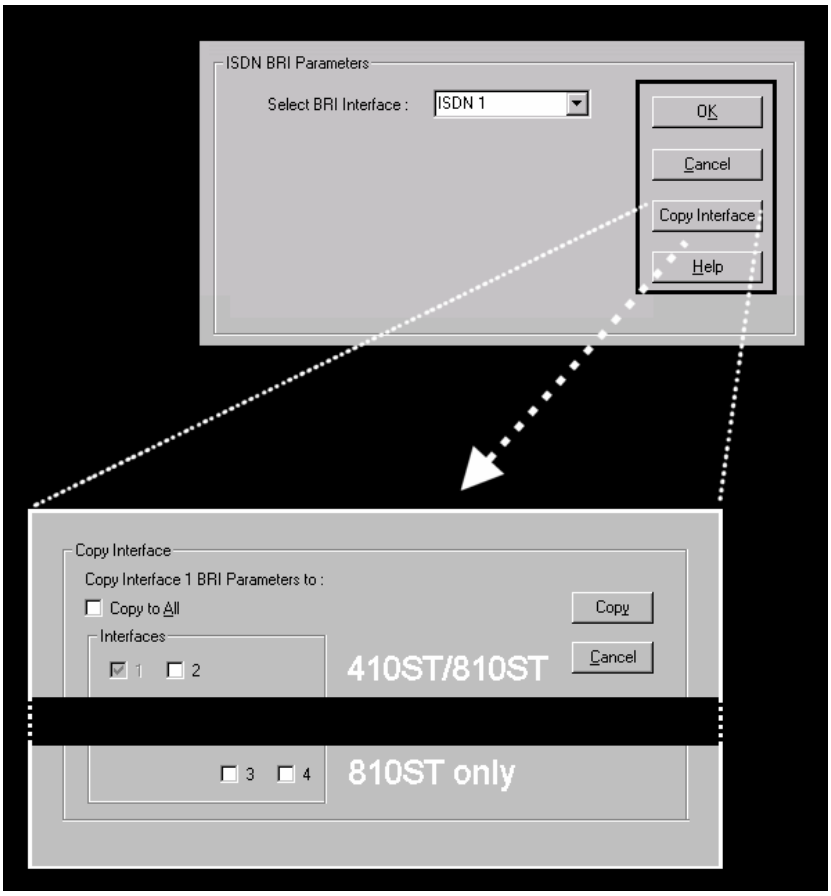
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td><code>Ctrl + F</code></td><td> 
</td></tr>
</table>

In the **ISDN BRI Parameters** screen, select one of the BRI interfaces and configure it for the particular implementation of ISDN that you will use. Configure each BRI interface per the requirements of your voip system. The MVP410ST has two ISDN-BRI interfaces and four channels; the MVP810ST has four ISDN-BRI interfaces and eight channels.



Note that ISDN BRI parameters are applied on an interface-by-interface basis. However, once you have established a set of ISDN BRI parameters for a particular interface, you can apply this entire set of parameters to another interface by using the **Copy Interface** button and its dialog box. To copy a set of ISDN BRI parameters to all interfaces, select "Copy to All" and click **Copy**.



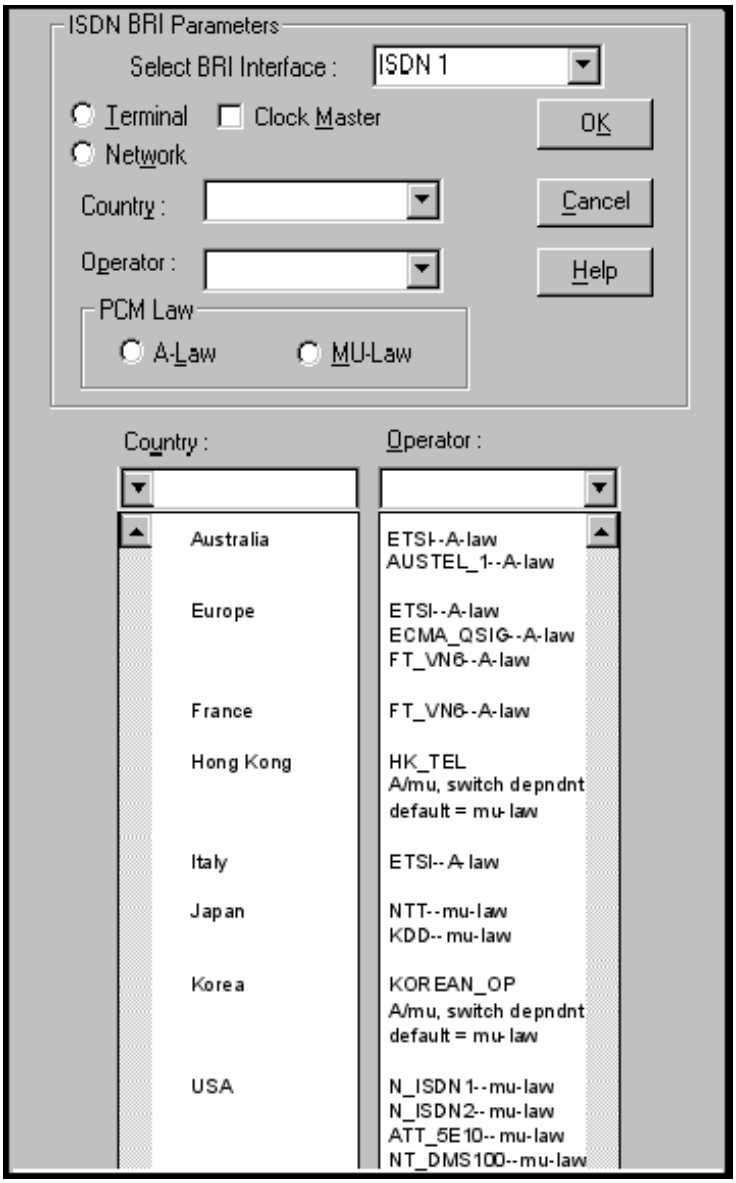
ISDN-BRI Parameter Definitions		
Field Name	**Values**	**Description**
Select BRI Interface	ISDN*n* for n= 1-2 (410ST) for n=1-4 (810ST)	In this field, you will choose which ISDN port you are configuring. The 410ST has two ISDN –BRI ports (or "interfaces"); the 810ST has four ISDN-BRI ports (or "interfaces"). Each port has two channels.
Layer 1 Interface	Terminal, Network, Clock Master	When "Terminal" is selected, it indicates that the MultiVOIP should emulate the subscriber (terminal) side of the digital connection. When "Network" is selected, it indicates that the MultiVOIP should emulate the central office (network) side of the digital connection. If connecting to a telco or PBX then choose "Terminal." If connecting to an ISDN phone or terminal adapter, then choose "Network." Default = Terminal. ISDN 1 Terminal Clock Master – The BRI VOIP uses an internal crystal as the clock master when any one of the ports is set for Network mode. If all ports are set for Terminal mode, then the first port to be activated becomes the clock master and it gets its clock from the network. The clock master can only be activated on ISDN1 interface in Terminal mode. If ISDN1 is selected and any other interface is set to Network mode, then clock master can be enabled. If enabled, ISDN1 is set as the clock master for all time, whether or not it is activated. This feature enables the BRI VOIP to have all of its interfaces synchronized to other network equipment and allows the terminal devices connected to the VOIP to be synchronized to the network equipment as well.

ISDN-BRI Parameter Definitions (continued)		
Field Name	**Values**	**Description**
Dialing Options	Inter Digit Timer (value in milliseconds)	Dialing options are relevant when the MultiVOIP provides dial tone either during an overlap receiving mode or providing a second dial tone. Default is 2000, which is 2 seconds. Range 250 ms to 10000 ms (1/4 sec to 10 sec).
Switch Information		
Country	see table below	Country in which MultiVOIP is operating with ISDN.
Operator	see table below	Indicates phone switch manufacturer/model or refers to telco so as to specify the switching system in question. ISDN is implemented somewhat differently in different switches (different software stacks are used).
PCM Law	A-Law or MU-Law	A-Law is an analog-to-digital compression /expansion standard used in Europe. MU-Law is the North America standard. Refer to the PCM-Law defaults based on country and operator below.
TEI *n* Assignment	Terminal or Network or both Automatic (Point-to-Point) Network – Automatic (Point-to-Multipoint) Fixed (Point-to-Point (0-63) TEI 0 & TEI 1 active	TEI (Terminal Endpoint Identifier) is a number to uniquely identify each device connected to the ISDN. TEI Assignment displays the value for each TEI assigned to the BRI port. Depending on the layer 1 interface selection (Terminal or Network) and the country selection, some fields are grayed out (inactive) as they have no meaning for this configuration. The TEI range is zero to 63 for Fixed and 64 to 126 for Automatic assignment. An interface in Network mode has the added option of Point-to-Multipoint Automatic TEI. This added option should be used in cases where the interface, or BRI port, has one or more terminal devices connected to it. This option may also help resolve some problems that occur when set for Point-to-Point Automatic TEI, even with only one terminal device connected to the interface. Point-to-Point Automatic TEI is then to be used when there is one network device and one terminal device, default setting. When selecting Fixed TEI, the connection must be Point-to-Point, one network and one terminal device.

ISDN-BRI Parameter Definitions (continued)		
Field Name	**Values**	**Description**
Numbering Details		
Calling Party	**Number Type:** Unknown, International, National, Net_Spf, Subscriber, Local, Abbreviated	Support for the user to select the Calling Party Number Type. Local is valid only for AT5 operator – local (directory) number. There may be cases where the default Type of Number and/or Numbering Plan may cause conflicts with the network or terminal device to which the VOIP is attached. Should this be the case, then changing to a different Type of Number might resolve the problem.
Called Party	**Number Type:** Unknown, International, National, Net_Spf, Subscriber, Local, Abbreviated **Number Plan:** Unknown, ISDN, Telephone, Data, Telex, National, Private, Reserved	Support for the user to select the Calling Party Number Type. Local is valid only for AT5 operator – local (directory) number. There may be cases where the default Type of Number and/or Numbering Plan may cause conflicts with the network or terminal device to which the VOIP is attached. Should this be the case, then changing to a different Type of Number might resolve the problem. Support for the user to select the Called Numbering Plan and Number Type. ISDN/telephony - CCITT E.164/E.163, Telephony – not in CEPT, Data – CCITT x.121, Telex – CCITT F.69, National – Standard, Reserved for extension. There may be cases where the default Type of Number and/or Numbering Plan may cause conflicts with the network or terminal device to which the VOIP is attached. Should this be the case, then changing to a different Type of Number might resolve the problem. Typically, setting the Called Party Type of Number to "unknown" and the Number Plan to "Unknown" will resolve such issues. These problems normally occur for calls going to the ISDN from the VOIP (numbers in the Inbound Phone Book)

ISDN-BRI Parameter Definitions (continued)		
Field Name	**Values**	**Description**
MSN Details		
MSN n	Right most digits to be distinguished	Multiple Subscriber Numbering (MSN) – In Euro-ISDN, and some country specific variants, it is possible to have several ISDN numbers for the same BRI or PRI connection. This feature allows you to assign different ISDN numbers to different applications, or different physical equipment, e.g., modem, phone instrument, fax machine, etc. When specifying MSN Numbers, don't enter the whole number, but only enough of the right most digits so that the numbers allocated to the line can be distinguished. Right now, a maximum of eight MSN Numbers is supported. For all ISDN incoming calls, when the dialed digits are matching with the configured MSN number, then, if the Dial Number corresponding to the matched MSN number is present, then using that Dial Number an IP outgoing call is made. Otherwise, for the matched MSN number, if no Dial Number is present, dial tone is provided to the user to dial the actual digits with which to make the IP outgoing call.
"Copy Interface" button		Copies the ISDN-BRI attributes of one interface to another interface. Attributes can be copied to multiple interfaces or to all interfaces at once. MSN details are not copied due to the nature of MSN assignment.

Country and Operator options for the MVP-410ST/810ST voip units are listed below.



11. **Set SNMP Parameters** (Remote Voip Management). *SNMP and MultiVoipManager are not supported for BRI MultiVOIPs. Please skip to step 12.*

    This dialog box can be reached by pulldown menu, keyboard shortcut, or sidebar. To make the MultiVOIP controllable by a remote PC running the MultiVoipManager software, check the "Enable SNMP Agent" box on the **SNMP Parameters** screen.

<table>
<tr><th colspan="2">Accessing "SNMP Parameters"</th></tr>
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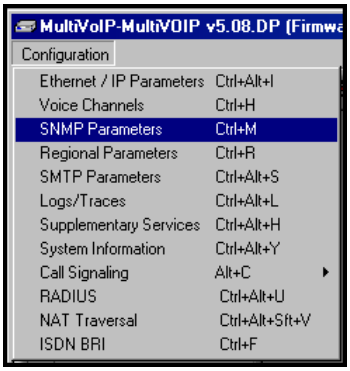
<tr><td> 
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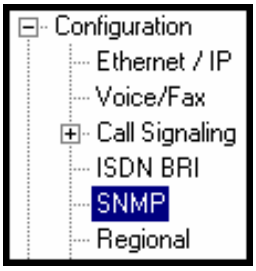
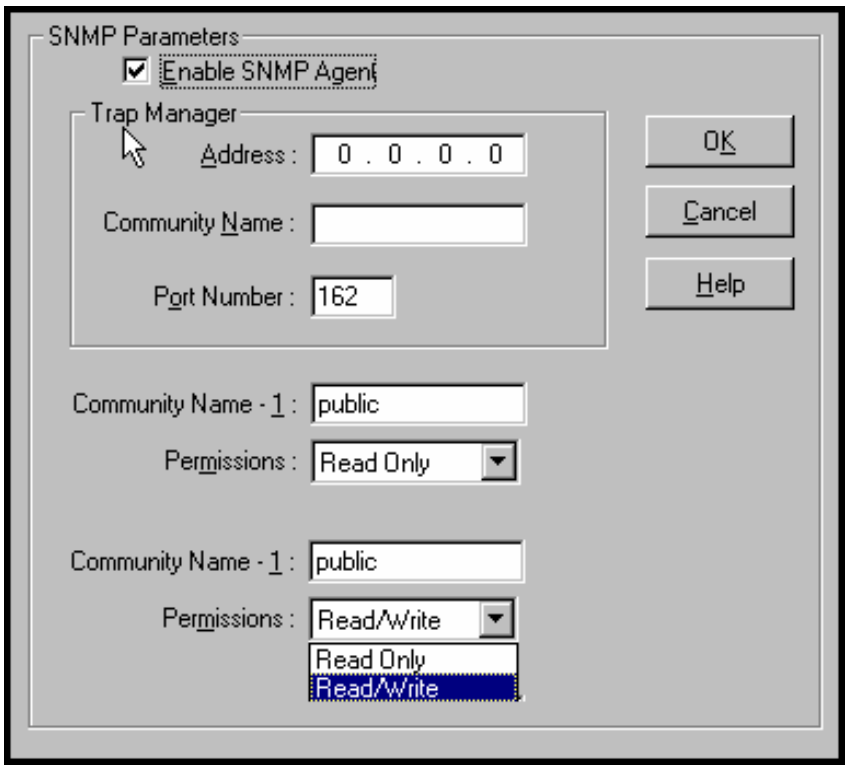
<tr><td>Ctrl + M</td><td> 
</td></tr>
</table>

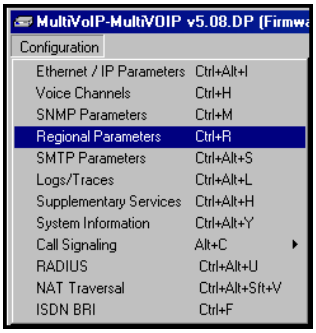
In each field, enter the values that fit your particular system.



The SNMP Parameter fields are described in the table below.

SNMP Parameter Definitions		
Field Name	**Values**	**Description**
Enable SNMP Agent	Y/N	Enables the SNMP code in the firmware of the MultiVOIP. This must be enabled for the MultiVOIP to communicate with and be controllable by the MultiVoipManager software. Default: disabled
Trap Manager Parameters		
Address	4 places; n.n.n.n n = 0-255	IP address of MultiVoipManager PC.
Community Name	--	A "community" is a group of VOIP endpoints that can communicate with each other. Often "public" is used to designate a grouping where all end users have access to entire VOIP network. However, calling permissions can be configured to restrict access as needed.
Port Number	162	The default port number of the SNMP manager receiving the traps is the standard port 162.
Community Name 1	Length = 19 characters (max.) Case sensitive.	First community grouping.
Permissions	Read-Only, Read/Write	If this community needs to change MultiVOIP settings, select Read/Write. Otherwise, select Read-Only to view settings.
Community Name 2	Length = 19 characters (max.) Case sensitive.	Second community grouping
Permissions	Read-Only, Read/Write	If this community needs to change MultiVOIP settings, select Read/Write. Otherwise, select Read-Only to view settings.

12. **Set Regional Parameters** (Phone Signaling Tones & Cadences). This dialog box can be reached by pulldown menu, keyboard shortcut, or sidebar.

<table>
<tr><th colspan="2">Accessing “Regional Parameters”</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td> 
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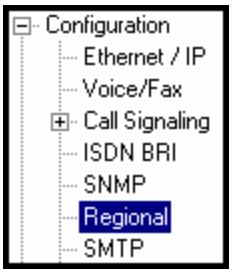
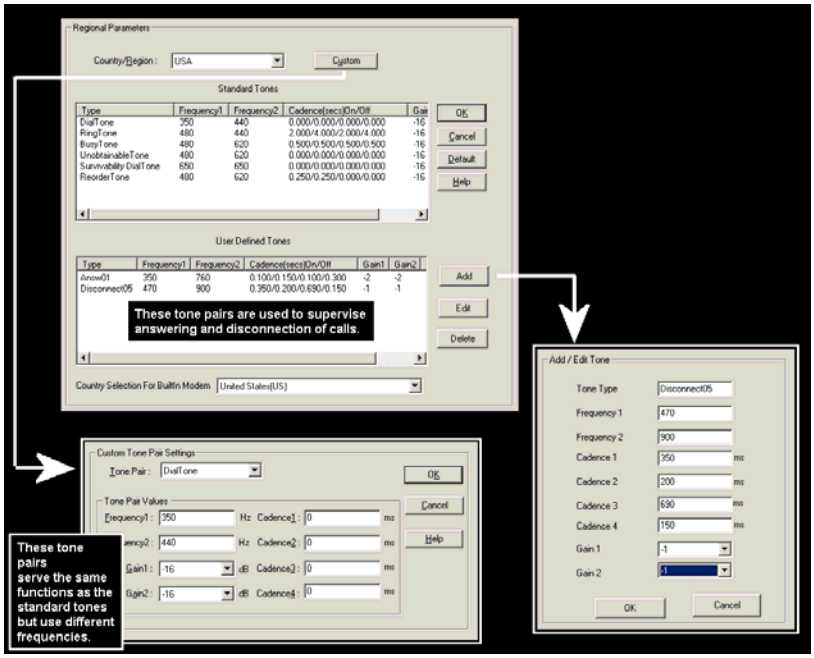
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>Ctrl + R</td><td> 
</td></tr>
</table>

The **Regional Parameters** screen will appear. For the country selected, the standard set of frequency pairs will be listed for dial tone, busy tone, 'unobtainable' tone (fast busy or trunk busy), ring tone, and other, more specialized tones.



In each field, enter the values that fit your particular system.

The **Regional Parameters** fields are described in the table below.

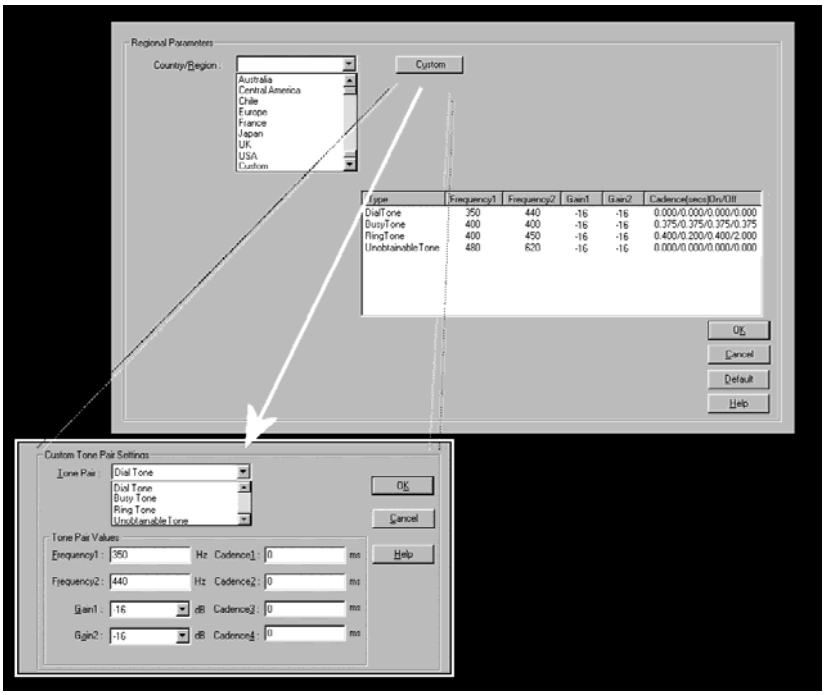
"Regional Parameter" Definitions		
Field Name	**Values**	**Description**
Country/ Region	USA, Japan, UK, Custom	Name of a country or region that uses a certain set of tone pairs for **dial tone, ring tone, busy tone**, **unobtainable tone** (fast busy tone), **survivability tone** (tone heard briefly, 2 seconds, after going offhook denoting survivable mode of VOIP unit), and **re-order tone** (a tone pattern indicating the need for the user to hang up the phone). In some cases, the tone-pair scheme denoted by a country name may also be used outside of that country. The "Custom" option (button) assures that any tone-pairing scheme worldwide can be accommodated.

"Regional Parameter" Definitions		
Field Name	**Values**	**Description**
Country/ Region	USA, Japan, UK, Custom **Note**: "Survivability" tone indicates a special type of call-routing redundancy.	Name of a country or region that uses a certain set of tone pairs for dial tone, ring tone, busy tone, and 'unobtainable' tone (fast busy tone), survivability tone (tone heard briefly, 2 seconds, after going offhook denoting survivable mode of voip unit) and re-order tone (a tone pattern indicating the need for the user to hang up the phone). In some cases, the tone-pair scheme denoted by a country name may also be used outside of that country. The "Custom" option (button) assures that any tone-pairing scheme worldwide can be accommodated.
Standard Tones fields		
Type column	dial tone, ring tone, busy tone, unobtainable tone (fast busy), survivability tone, re-order tone	Type of telephony tone-pair for which frequency, gain, and cadence are being presented.
Frequency 1	freq. in Hertz	Lower frequency of pair.
Frequency 2	freq. in Hertz	Higher frequency of pair.

"Regional Parameter" Definitions (cont'd)		
Field Name	**Values**	**Description**
Standard Tones fields (cont'd)		
Gain 1	gain in dB +3dB to –31dB and "mute" setting	Amplification factor of lower frequency of pair. This applies to the dial, ring, busy and 'unobtainable' tones that the MultiVOIP outputs as audio to the ISDN B-channel port. **Default: -16dB**
Gain 2	gain in dB +3dB to –31dB and "mute" setting	Amplification factor of higher frequency of pair. This applies to the dial, ring, busy, and 'unobtainable' (fast busy) tones that the MultiVOIP outputs as audio to the ISDN B-channel port. **Default: -16dB**
Cadence (msec) On/Off	n/n/n/n four integer time values in milli-seconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote phone ringing, phone busy, connection unobtainable (fast busy), dial tone ("0" indicates continuous tone), survivability, and re-order. Default values differ for different countries/regions. Although most cadences have only two parts (an "on" duration and an "off" duration), some telephony cadences have four parts. Most cadences, then, are expressed as two iterations of a two-part sequence. Although this is redundant, it is necessary to allow for expression of 4-part cadences.
Custom (button)	--	Click on the "Custom" button to bring up the **Custom Tone Pair Settings** screen. (The "Custom" button is active only when "Custom" is selected in the **Country/Region** field.) This screen allows the user to specify tone pair attributes that are not found in any of the standard national/regional telephony toning schemes.

"Regional Parameter" Definitions (cont'd)		
Field Name	**Values**	**Description**
User Defined Tones fields		
Type column	alphanumeric name specified by user	Name of supervisory tone pair. Cannot be same as name of any standard tone pair.
Frequency 1	freq. in Hertz	Lower frequency of pair.
Frequency 2	freq. in Hertz	Higher frequency of pair.
Gain 1	gain in dB +3dB to –31dB and "mute" setting	Amplification factor of lower frequency of pair. This applies to any supervisory tones that the MultiVOIP outputs as audio to the ISDN B-channel port. **Default: -16dB**
Gain 2	gain in dB +3dB to –31dB and "mute" setting	Amplification factor of higher frequency of pair. This applies to any supervisory tones that the MultiVOIP outputs as audio to the ISDN B-channel port. **Default: -16dB**
Cadence (msec) On/Off	n/n/n/n four integer time values in milli-seconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote supervisory tones specified by user. Supervisory tones relate to answering and disconnection of calls. Although most cadences have only two parts (an "on" duration and an "off" duration), some telephony cadences have four parts. Most cadences, then, are expressed as two iterations of a two-part sequence. Although this is redundant, it is necessary to allow for expression of 4-part cadences.

13. **Set Custom Tones and Cadences** (optional). The **Regional Parameters** dialog box has a secondary dialog box that allows you to customize DTMF tone pairs to create unique ring-tones, dial-tones, busy-tones or "unobtainable" tones (fast busy signal) or "re-order" tones (telling the user that she must hang up an off-hook phone) or "survivability" tones (an indication of call-routing redundancy) for your system. This screen allows the user to specify tone-pair attributes that are not found in any of the standard national/regional telephony toning schemes. To access this customization feature, click on the **Custom** button on the **Regional Parameters** screen. (The "Custom" button is active only when "Custom" is selected in the **Country/Region** field.)



The **Custom Tone-Pair Settings** fields are described in the table below.

Custom Tone-Pair Settings Definitions		
Field Name	**Values**	**Description**
Tone Pair	dial tone, busy tone, ring tone, 'unobtainable' tone, survivability tone, re-order tone	Identifies the type of telephony signaling tone for which frequencies are being specified.
TONE PAIR VALUES		**About Defaults**: US telephony values are used as defaults on this screen. However, since this dialog box is provided to allow custom tone-pair settings, default values are essentially irrelevant.
Frequency 1	frequency in Hertz	Frequency of lower tone of pair. This outbound tone pair enters the MultiVOIP at the input port.
Frequency 2	frequency in Hertz	Frequency of higher tone of pair. This outbound tone pair enters the MultiVOIP at the input port.
Gain 1	gain in dB +3dB to –31dB and "mute" setting	Amplification factor of lower frequency of pair. This figure describes amplification that the MultiVOIP applies to outbound tones entering the MultiVOIP at the input port. **Default = -16dB**
Gain 2	gain in dB +3dB to –31dB and "mute" setting	Amplification factor of higher frequency of pair. This figure describes amplification that the MultiVOIP applies to outbound tones entering the MultiVOIP at the input port. **Default = -16dB**

Custom Tone-Pair Settings Definitions		
Field Name	**Values**	**Description**
Cadence 1	integer time value in milli-seconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote phone ringing, phone busy, dial tone ("0" indicates continuous tone) survivability and re-order. Cadence 1 is duration of first period of tone being "on" in the cadence of the telephony signal (which could be ring-tone, busy-tone, unobtainable-tone, or dial tone).
Cadence 2	duration in milliseconds	Cadence 2 is duration of first "off" period in signaling cadence.
Cadence 3	duration in milliseconds	Cadence 3 is duration of second "on" period in signaling cadence.
Cadence 4	duration in milliseconds	Cadence 4 is duration of second "off" period in the signaling cadence, after which the 4-part cadence pattern of the telephony signal repeats.

14. **Set SMTP Parameters** (Log Reports by Email). The **SMTP Parameters** screen is applicable when the VOIP administrator has chosen to receive log reports by email (this is done by selecting the "SMTP" checkbox in the **Logs/Traces** screen and selecting "Enable SMTP" in the **SMTP Parameters** screen.). The **SMTP Parameters** screen can be reached by pulldown menu, keyboard shortcut, or sidebar.

<table>
<tr><th colspan="2">Accessing "SMTP Parameters"</th></tr>
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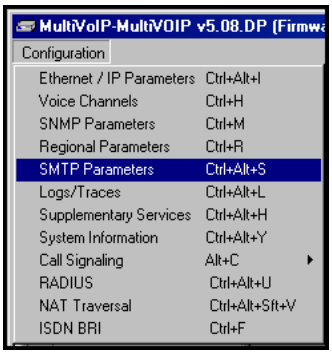
<tr><td> 
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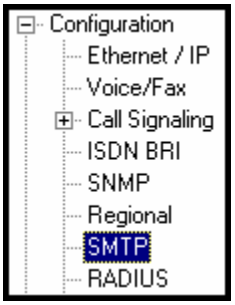
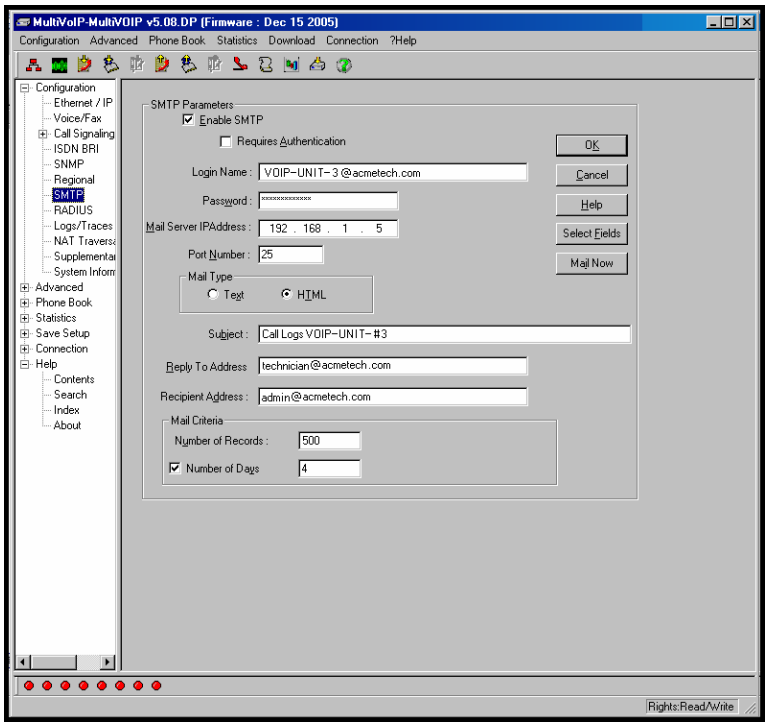
<tr><td colspan="2"></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>Ctrl + Alt + S</td><td> 
</td></tr>
</table>

MultiVOIP as Email Sender. When SMTP is used, the MultiVOIP will actually be given its own email account (with Login Name and Password) on some mail server connected to the IP network. Using this account, the MultiVOIP will then send out email messages containing log report information. The "Recipient" of the log report email is ordinarily the VoIP administrator. Because the MultiVOIP cannot receive email, a "Reply-To" address must also be set up. Ordinarily, the "Reply-To" address is that of a technician who has access to the mail server or MultiVOIP or both, and the VoIP administrator might also be designated as the "Reply-To" party. The main function of the Reply-To address is to receive error or failure messages regarding the emailed reports.

The **SMTP Parameters** screen is shown below.

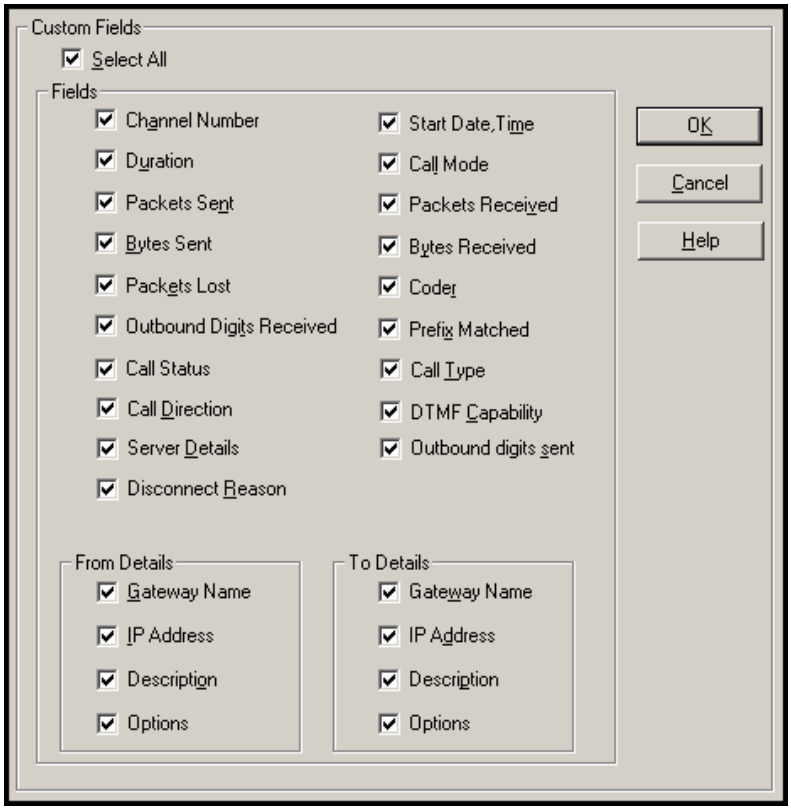


"SMTP Parameters" Definitions		
Field Name	**Values**	**Description**
Enable SMTP	Y/N	In order to send log reports by email, this box must be checked. However, to enable SMTP functionality, you must also select "SMTP" in the **Logs** screen.
Requires Authentication	Y/N	If this checkbox is checked, the MultiVOIP will send Authentication information to the SMTP server. The authentication information indicates whether or not the email sender has permission to use the SMTP server.
Login Name	alpha-numeric, per email domain	This is the User Name for the MultiVOIP unit's email account.

.

"SMTP Parameters" Definitions (cont'd)		
Field Name	**Values**	**Description**
Password	alpha-numeric	Login password for MultiVOIP unit's email account.
Mail Server IP Address	n.n.n.n for n= 0 to 255	This is the mail server's IP address. This mail server must be accessible on the IP network to which the MultiVOIP is connected.
Port Number	25	25 is a standard port number for SMTP.
Mail Type	text or html	Mail type in which log reports will be sent.
Subject	text	User specified. Subject line that will appear for all emailed log reports for this MultiVOIP unit.
Reply-To Address	email address	User specified. This email address functions as a source email identifier for the MultiVOIP, which, of course, cannot usefully receive email messages. The Reply-To address provides a destination for returned messages indicating the status of messages sent by the MultiVOIP (esp. to indicate when log report email was undeliverable or when an error has occurred).
Recipient Address	email address	User specified. Email address at which VOIP administrator will receive log reports.
Mail Criteria		Criteria for sending log summary by email. The log summary email will be sent out either when the user-specified number of log messages has accumulated, or once every day or multiple days, *which ever comes first*.
Number of Records	integer	This is the number of log records that must accumulate to trigger the sending of a log-summary email.
Number of Days	integer	This is the number of days that must pass before triggering the sending of a log-summary email.

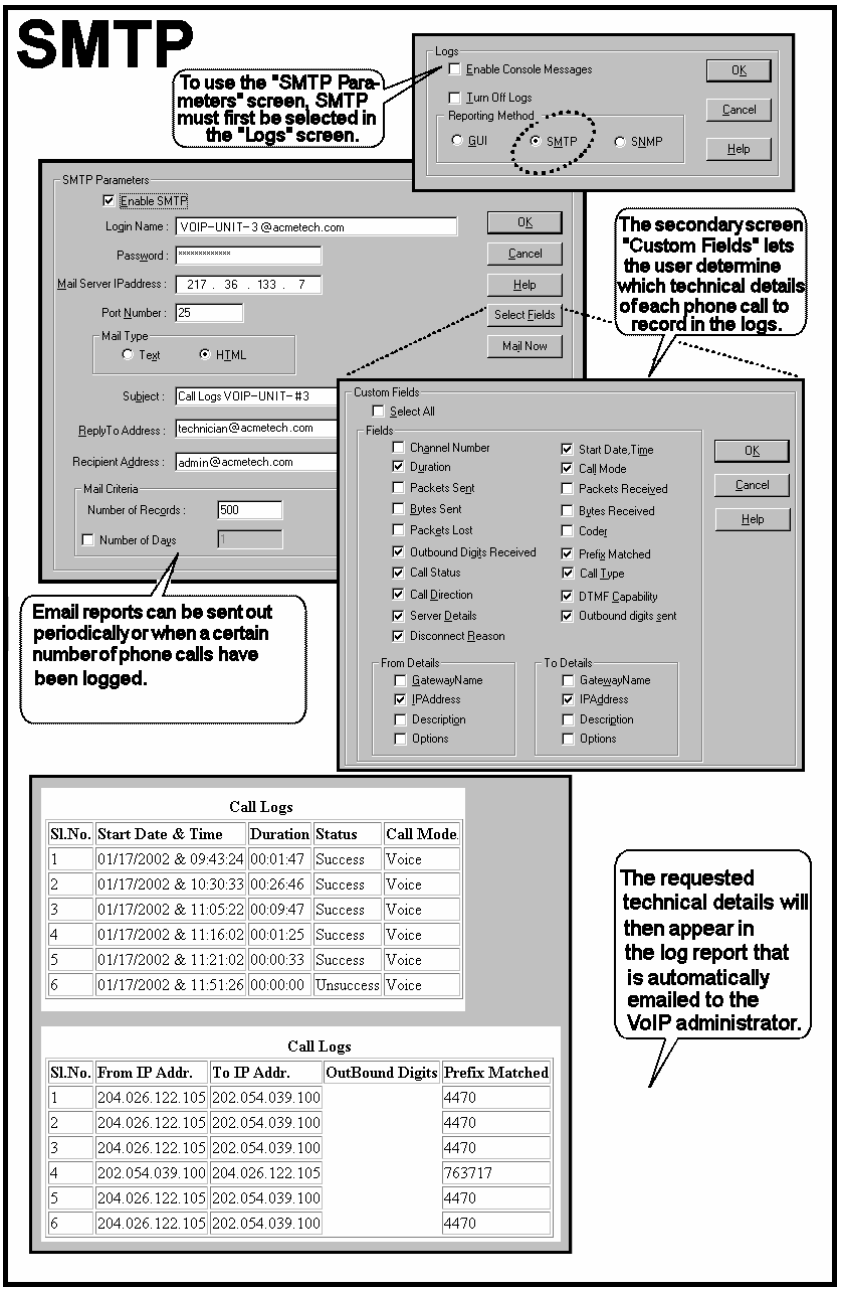
The **SMTP Parameters** dialog box has a secondary dialog box, **Custom Fields**, that allows you to customize email log messages for the MultiVOIP. The MultiVOIP software logs data about many aspects of the call traffic going through the MultiVOIP. The Custom Fields screen lets you pick which aspects will be included in the email log reports.



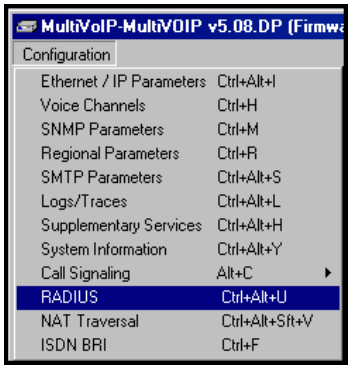
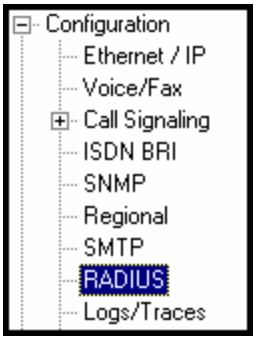
"Custom Fields" Definitions			
Field	**Description**	**Field**	**Description**
Select All	Log report to include all fields shown.		
Channel Number	Data channel carrying call.	Start Date, Time	Date and time the phone call began.
Duration	Length of call.	Call Mode	Voice or fax.
Packets Sent	Total packets sent in call.	Packets Received	Total packets received in call.

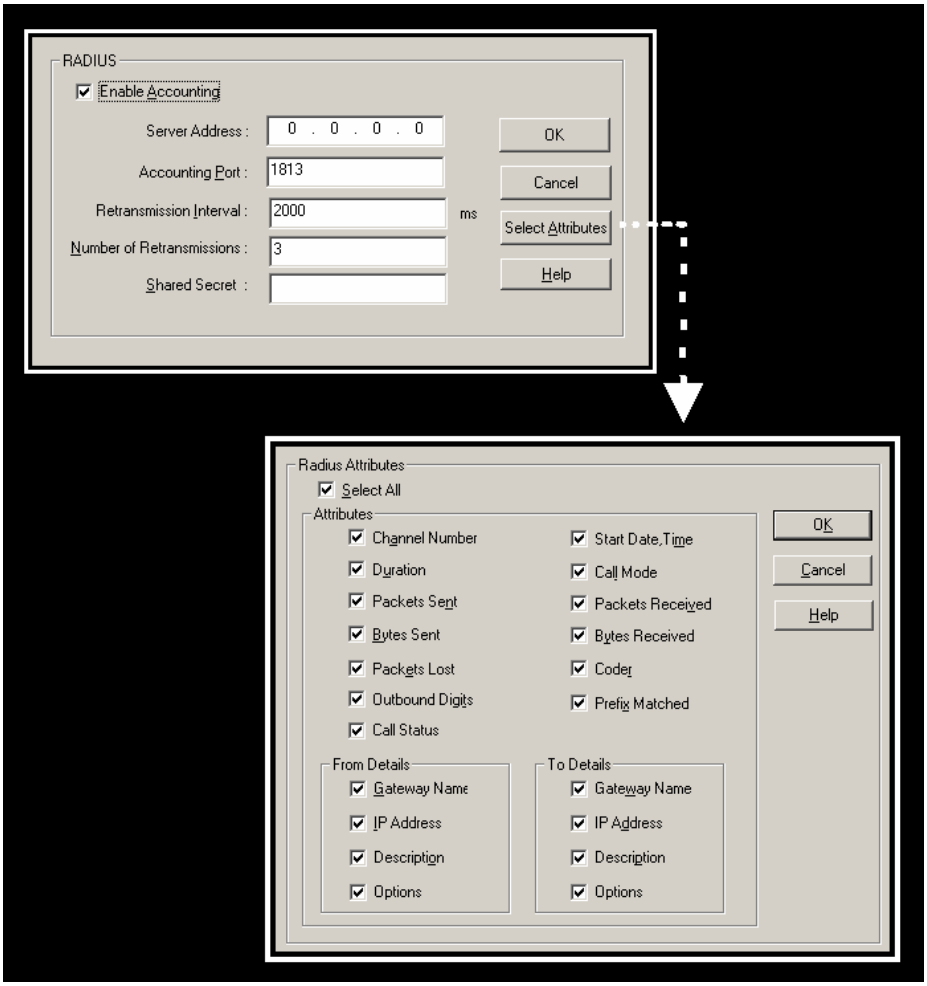
"Custom Fields" Definitions (cont'd)			
Field	**Description**	**Field**	**Description**
Bytes Sent	Total bytes sent in call.	Bytes Received	Total bytes received in call.
Packets Lost	Packets lost in call.	Coder	Voice Coder /Compression Rate used for call will be listed in log.
Outbound Digits Received	The DTMF dialing digits received by this gateway from the remote gateway presuming that DTMF is set to "Out of Band."	Prefix Matched	When selected, the phonebook prefix matched in processing the call will be listed in log.
Call Status	Successful or unsuccessful.	Call Type	Indicates the Call Signaling protocol used for the call (H.323, SIP, or SPP).
Call Direction	Indicates call's originating party.	DTMF Capability	Indicates whether the DTMF dialing digits are carried "Inband" or "Out of Band." The corresponding field values differ for the 3 different voip protocols. For H.323, this field can display "Out of Band" or "Inband". For SIP it can display either "Out of Band RFC2833" or "Out of Band SIP INFO" to indicate the out-of-band condition or "Inband" to indicate the in-band condition. For SPP it can display "Out of Band RFC2833" or "Inband".

"Custom Fields" Definitions (cont'd)			
Field	**Description**	**Field**	**Description**
Server Details	The IP address of the traffic control server (if any) being used (whether an H.323 gatekeeper, a SIP proxy, or an SPP registrar gateway) will be displayed here if the call is handled through that server.	Outbound Digits Sent	The dialing digits sent by this gateway to the remote gateway presuming that DTMF is set to "Out of Band."
Disconnect Reason	Indicates whether the call was disconnected simply because the desired conversation was done or some other irregular cause occasioned disconnection (e.g., a technical error or failure). Values are "Normal" and "Local" disconnection.		
From Details		**To Details**	
Gateway Number	Originating gateway	Gatew N.	Completing or answering gateway
IP Addr	IP address where call originated.	IP Addr	IP address where call was completed or answered.
Descript	Identifier of site where call originated.	Descript	Identifier of site where call was completed or answered.
Options	When selected, log will not Silence Compression and Forward Error Correction by call originator.	Options	When selected, log will not use Silence Compression and Forward Error Correction by party answering call.


SMTP
To use the "SMTP Parameters" screen, SMTP must first be selected in the "Logs" screen.
Logs
Enable Console Messages
Turn Off Logs
Reporting Method
GUI
SMTP
SNMP
OK
Cancel
Help
SMTP Parameters
Enable SMTP
Login Name : VOIP-UNIT-3 @acmetech.com
Password :
Mail Server IPaddress : 217 . 36 . 133 . 7
Port Number : 25
Mail Type
Text
HTML
Select Fields
Mail Now
Subject : Call Logs VOIP-UNIT-#3
ReplyTo Address : technician@acmetech.com
Recipient Address : admin@acmetech.com
Mail Criteria
Number of Records : 500
Number of Days
The secondary screen "Custom Fields" lets the user determine which technical details of each phone call to record in the logs.
Custom Fields
Select All
Fields
Channel Number
Duration
Packets Sent
Bytes Sent
Packets Lost
Outbound Digits Received
Call Status
Call Direction
Server Details
Disconnect Reason
Start Date,Time
Call Mode
Packets Received
Bytes Received
Coder
Prefix Matched
Call Type
DTMF Capability
Outbound digits sent
From Details
GatewayName
IPAddress
Description
Options
To Details
GatewayName
IPAddress
Description
Options
Email reports can be sent out periodically or when a certain number of phone calls have been logged.
Call Logs
Sl.No. Start Date & Time Duration Status Call Mode
1 01/17/2002 & 09:43:24 00:01:47 Success Voice
2 01/17/2002 & 10:30:33 00:26:46 Success Voice
3 01/17/2002 & 11:05:22 00:09:47 Success Voice
4 01/17/2002 & 11:16:02 00:01:25 Success Voice
5 01/17/2002 & 11:21:02 00:00:33 Success Voice
6 01/17/2002 & 11:51:26 00:00:00 Unsuccess Voice
Call Logs
Sl.No. From IP Addr. To IP Addr. OutBound Digits Prefix Matched
1 204.026.122.105 202.054.039.100 4470
2 204.026.122.105 202.054.039.100 4470
3 204.026.122.105 202.054.039.100 4470
4 202.054.039.100 204.026.122.105 763717
5 204.026.122.105 202.054.039.100 4470
6 204.026.122.105 202.054.039.100 4470
The requested technical details will then appear in the log report that is automatically emailed to the VoIP administrator.

15. **Set RADIUS parameters**. You can access the RADIUS screen by pulldown menu, sidebar menu, or keyboard shortcut. In general, RADIUS is concerned with authentication, authorization, and accounting. The MultiVOIP BRI units support only the accounting function. The accounting function is well suited for billing of voip telephony services. In the **Attributes** secondary screen (accessed by clicking on Select Attributes), the voip administrator can select the parameters to be tallied by the RADIUS server.

<table>
<tr><th colspan="2">Accessing “RADIUS” Parameters</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td> 
</td><td>--</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>Ctrl + Alt + U</td><td> 
</td></tr>
</table>



The fields of the RADIUS screen are described in the table below.

RADIUS Screen Field Definitions		
Field Name	**Values**	**Description**
Enable Accounting	Y/N	When checked, the MultiVOIP will access the accounting functionality of the
Server Address	n.n.n.n 0 – 255	IP address of the RADIUS server that handles accounting (billing) for the current MultiVOIP unit.
Accounting Port	numeric; 1 - 65535	TDM time slot at which RADIUS accounting information will be transmitted and received.
Retrans-mission Interval		If the MultiVOIP sends out a packet to the RADIUS server and doesn't receive a response in the retransmit interval, it will retransmit that packet again and wait the retransmit interval again for a response. How many times it does this is determined by the setting in the **Number of Retransmissions** field.
Number of Re-transmis-sions	0 - 255	
Shared Secret	alpha-numeric	Client encryption key for the current voip unit.
Select Attributes (button)	--	Gives access to RADIUS Attributes screen. On Attributes screen, one can specify the parameters to be tallied by the RADIUS server for accounting (usually billing) purposes.

The **RADIUS Parameters** dialog box has a secondary dialog box, **RADIUS Attributes**, that allows you to customize accounting information sent to the RADIUS server by the MultiVOIP. The MultiVOIP software logs data about many aspects of the call traffic going through the MultiVOIP. The RADIUS Attributes screen lets you pick which aspects will be included in the accounting reports sent to the RADIUS server.

"RADIUS Attributes" Field Definitions			
Field	**Description**	**Field**	**Description**
Select All	Log report to include all fields shown.		
Channel Number	Data channel carrying call.	Start Date, Time	Date and time the phone call began.
Duration	Length of call.	Call Mode	Voice or fax.
Packets Sent	Total packets sent in call.	Packets Received	Total packets received in call.
Bytes Sent	Total bytes sent in call.	Bytes Received	Total bytes received in call.
Packets Lost	Packets lost in call.	Coder	Voice Coder /Compression Rate used for call will be listed in log.
Outbound Digits Sent	The DTMF dialing digits received by this gateway from the remote gateway presuming that DTMF is set to "Out of Band."	Prefix Matched	When selected, the phonebook prefix matched in processing the call will be listed in log.
Call Status	Successful or unsuccessful.		
Server Details	The IP address (etc.) of the traffic control server (if any) being used (whether an H.323 gatekeeper, a SIP proxy, or an SPP registrar gateway) will be displayed here if the call is handled through that server. The Options field refers to non-mandatory server features that might be activated. For example, with H.323, various H.323 Version 4 options might be listed (Multiplexing, Tunneling, etc.).		

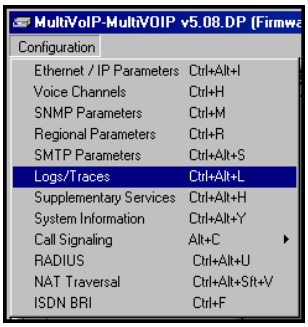
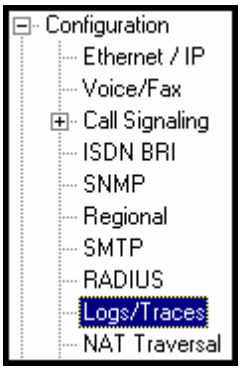
"RADIUS Attributes" Field Definitions (cont'd)			
Field	**Description**	**Field**	**Description**
From Details		**To Details**	
Gateway Number	Originating gateway	Gatew N.	Completing or answering gateway
IP Addr	IP address where call originated.	IP Addr	IP address where call was completed or answered.
Descript	Identifier of site where call originated.	Descript	Identifier of site where call was completed or answered.
Options	When selected, log will not use Silence Compression and Forward Error Correction by call originator.	Options	When selected, log will not use Silence Compression and Forward Error Correction by party answering call.

16. **Set Log Reporting Method**. You can access the **Logs/Traces** screen by pulldown menu, sidebar menu, or keyboard shortcut. The Logs/Traces screen lets you choose how the VoIP administrator will receive log reports about the MultiVOIP's performance and the phone call traffic that is passing through it. Log reports can be received in one of three ways:

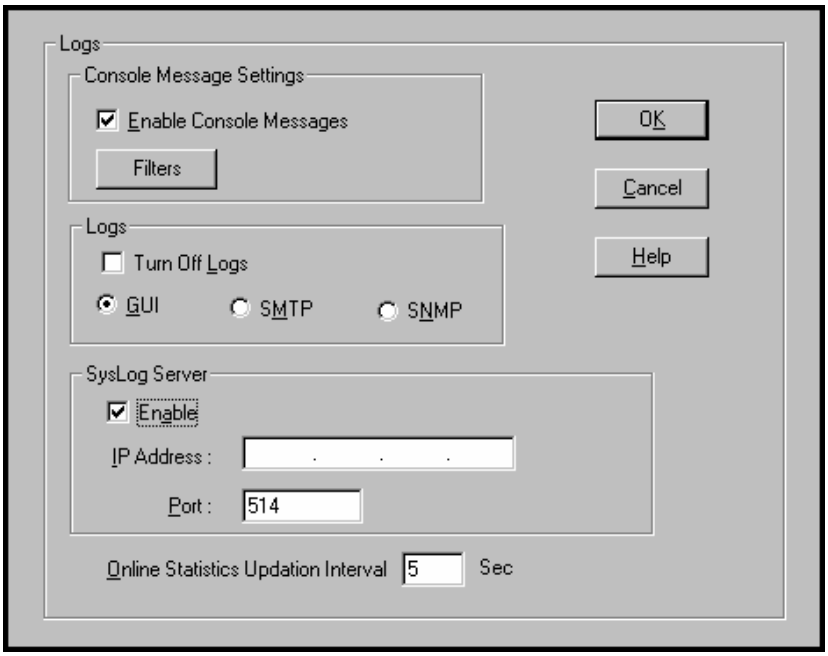
    A. in the MultiVOIP program (GUI),

    B. via email (SMTP), or

    C. at the MultiVoipManager remote voip system management program (SNMP).

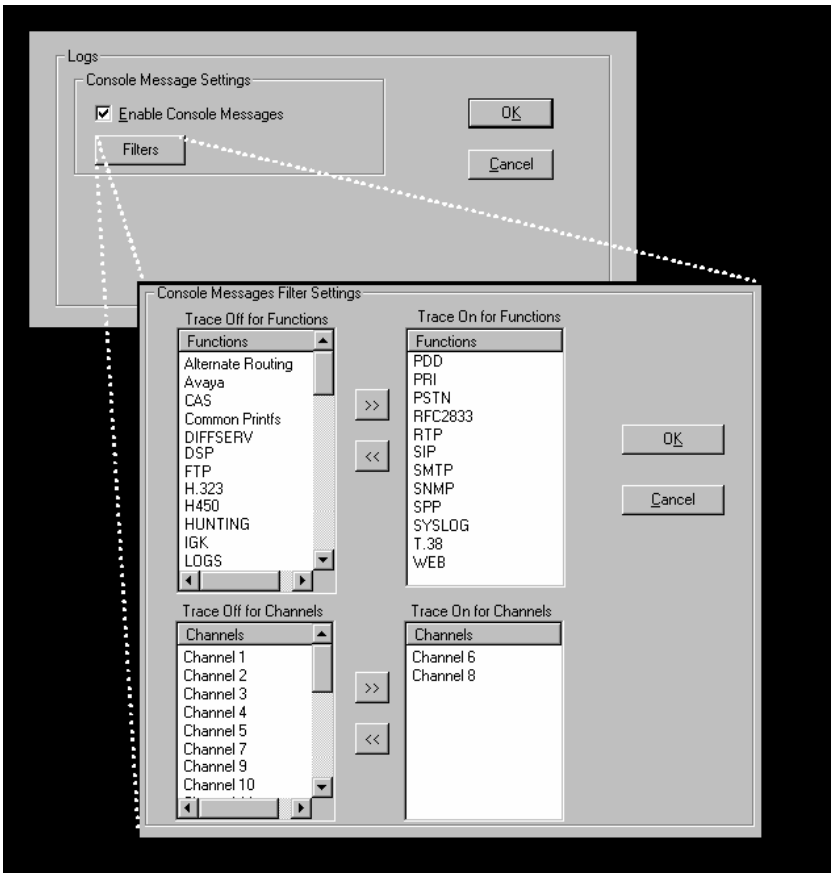
<table>
<tr><th colspan="2">Accessing "Logs/Traces" Screen</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td> 
</td><td></td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>Ctrl + Alt + L</td><td> 
</td></tr>
</table>

If you enable console messages, you can customize the types of messages to be included/excluded in log reports by clicking on the "Filters" button and using the **Console Messages Filter Settings** screen (see subsequent page). If you use the logging function, select the logging option that applies to your VoIP system design. If you intend to use a SysLog Server program for logging, click in that Enable check box. The common SysLog logical port number is 514. If you intend to use the MultiVOIP web browser GUI for configuration and control of MultiVOIP units, be aware that the web browser GUI does not support logs directly. However, when the web browser GUI is used, log files can still be sent to the voip administrator via email (which requires activating the SMTP logging option in this screen).

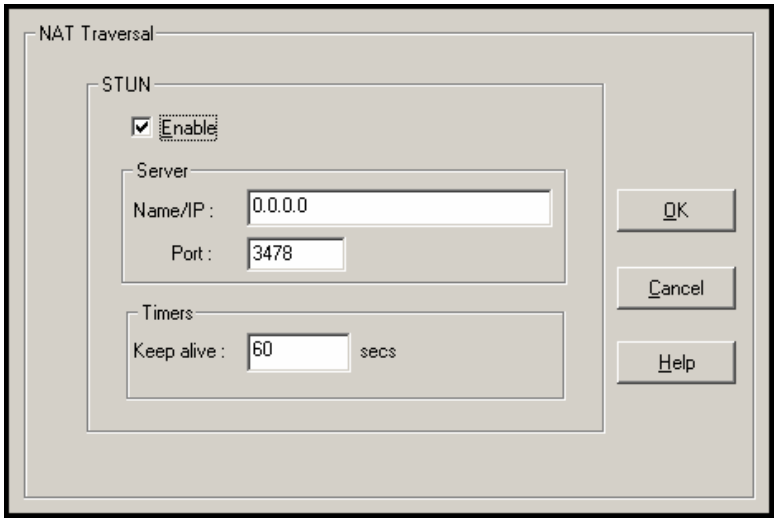


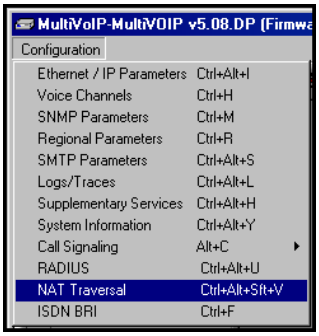
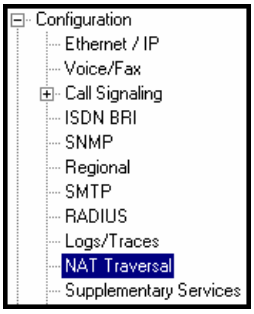
"Logs" Screen Definitions		
Field Name	**Values**	**Description**
Enable Console Messages	Y/N	Allows MultiVOIP debugging messages to be read via a basic terminal program like HyperTerminal ™ or equivalent. Normally, this should be disabled because it uses MultiVOIP processing resources. Console messages are meant for tech support personnel.
Filters (button)		Click to access secondary screen on where console messages can be included/excluded by category and on a per-channel basis. (See the Console Messages Filter Settings screen on subsequent page.)
Turn Off Logs	Y/N	Check to disable log-reporting function.
Logs Buttons		Only one of these three log reporting methods, GUI, SMTP, or SNMP, may be chosen.
GUI	Y/N	User must view logs at the MultiVOIP configuration program.
SNMP	Y/N	Log messages will be delivered to the MultiVoipManager application program.
SMTP	Y/N	Log messages will be sent to user-specified email address.
SysLog Server Enable	Y/N	This box must be checked if logging is to be done in conjunction with a SysLog Server program. For more on SysLog Server, see *Operation & Maintenance* chapter.
IP Address	n.n.n.n for n= 0-255	IP address of computer, connected to voip network, on which SysLog Server program is running.
Port	514	Logical port for SysLog Server. 514 is commonly used.
Online Statistics Updation Interval	integer	Set the interval (in seconds) at which logging information will be updated.

To customize console messages by category and/or by channel, click on "Filters" and use the **Console Messages Filters Settings** screen.



17. **Set NAT Traversal** parameters. You can access the NAT Traversal screen by pulldown menu, sidebar menu, or keyboard shortcut. NAT (Network Address Translation) parameters are applicable only when the MultiVOIP is operating in SIP mode. The use of STUN (Simple Traversal of UDP NATs) servers to aid networks with NAT devices is described in RFC 3489.



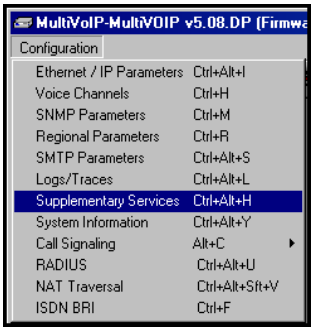
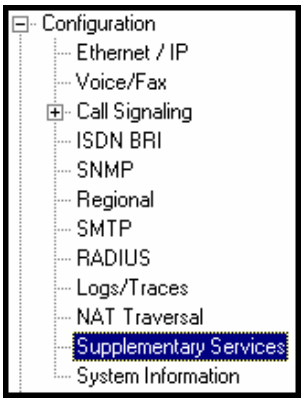
Accessing "NAT Traversal" Parameters	
Pulldown	**Sidebar**
	
Shortcut	**Icon**
Ctrl + Alt + Sft + V	--

Descriptions for NAT Traversal screen fields are presented in the table below.

NAT Traversal Definitions (cont'd)		
Field Name	**Values**	**Description**
Enable (STUN)	Y/N	Enables STUN client functionality in the MultiVOIP. STUN (Simple Traversal of UDP through NATs (Network Address Translation)) is a protocol that allows a server to assist client gateways behind a NAT firewall or router with their packet routing.
Name/IP (Server)	n.n.n.n 0 - 255	IP address of the STUN server.
Port (Server; NAT/STUN)	numeric; default= 3478	The data port (TDM time slot) at which STUN info will be transmitted and received.
Keep Alive (Timers; NAT/STUN)	60 – 3600 (in seconds)	The interval at which the STUN client sends indicator ("Keep Alive") packets to the STUN server to determine whether or not the STUN server is available.

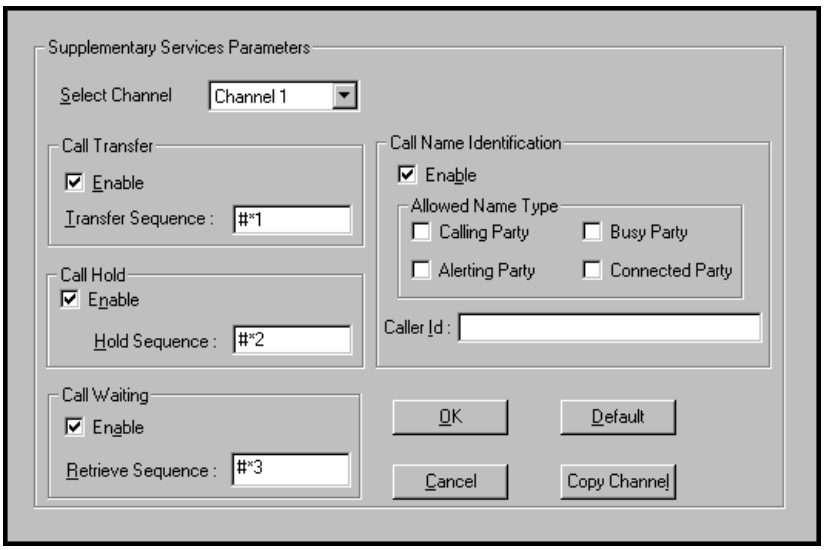
18. **Set Supplementary Services Parameters.** This dialog box can be reached by pulldown menu, keyboard shortcut, or sidebar.

Accessing "Supplementary Services" Parameters

Pulldown	Icon
	
Shortcut	**Sidebar**
`Ctrl + Alt +H`	

Supplementary Services features derive from the H.450 standard, which brings to voip telephony functionality once only available with PSTN or PBX telephony. Supplementary Services features can be used under H.323 only and *not* under SIP. Even though the H.450 standard refers only to H.323, Supplementary Services are still applicable to the SIP and SPP voip protocols.

In each field, enter the values that fit your particular network.



Of the features implemented under Supplementary Services, three are very closely related: Call Transfer, Call Hold, and Call Waiting. Call Name Identification is similar but not identical to the premium PSTN feature commonly known as **Caller ID**.

Call Transfer. Call Transfer allows one party to re-connect the party with whom they have been speaking to a third party. The first party is disconnected when the third party becomes connected. Feature is invoked by a programmable phone keypad sequence (for example, #7).

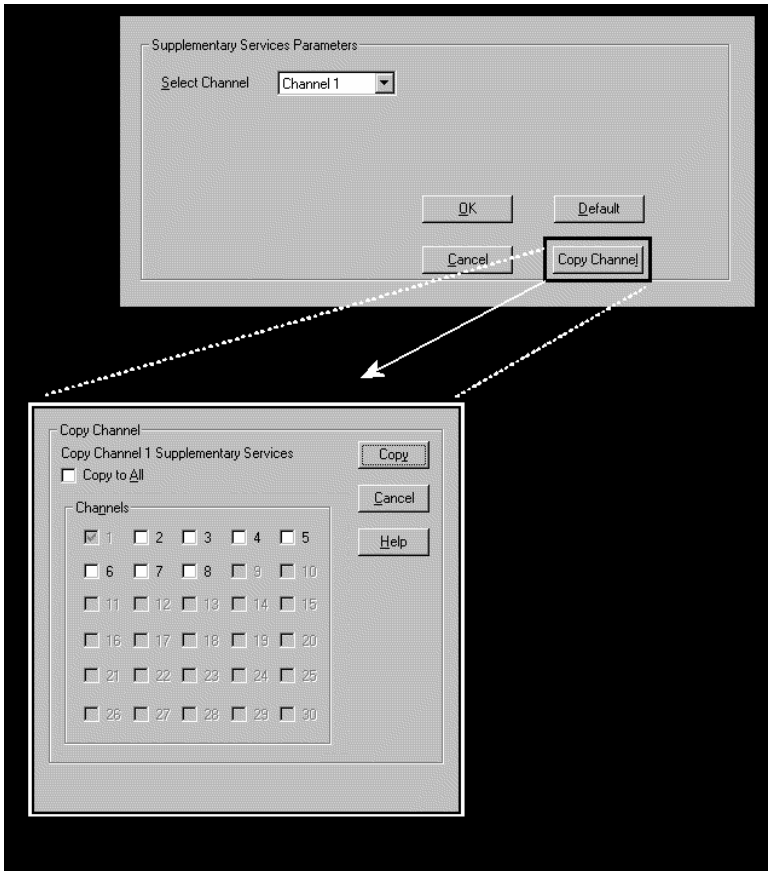
Call Hold. Call Hold allows one party to maintain an idle (non-talking) connection with another party while receiving another call (Call Waiting), while initiating another call (Call Transfer), or while performing some other call management function. Invoked by keypad sequence.

Call Waiting. Call Waiting notifies an engaged caller of an incoming call and allows them to receive a call from a third party while the party with whom they have been speaking is put on hold. Invoked by keypad sequence.

Call Name Identification. When enabled for a given voip unit (the 'home' voip), this feature gives notice to remote voips involved in calls. Notification goes to the remote voip administrator, not to individual phone stations. When the home voip is the caller, a plain English descriptor will be sent to the remote (callee) voip identifying

the channel over which the call is being originated (for example, "Calling Party - Omaha Sales Office Line 2"). If that voip channel is dedicated to a certain individual, the descriptor could say that, as well (for example "Calling Party - Harold Smith in Omaha"). When the home voip receives a call from any remote voip, the home voip sends a status message back to that caller. This message confirms that the home voip's phone channel is either busy or ringing or that a connection has been made (for example, "Busy Party - Omaha Sales Office Line 2"). These messages appear in the **Statistics – Call Progress** screen of the remote voip.

Note that Supplementary Services parameters are applied on a channel-by-channel basis. However, once you have established a set of supplementary parameters for a particular channel, you can apply this entire set of parameters to another channel by using the **Copy Channel** button and its dialog box. To copy a set of Supplementary Services parameters to all channels, select "Copy to All" and click **Copy**.



The **Supplementary Services** fields are described in the tables below.

Supplementary Services Parameter Definitions		
Field Name	**Values**	**Description**
Select Channel	1-4 (410ST) 1-8 (810ST)	The channel to be configured is selected here.
Call Transfer Enable	Y/N	Select to enable the Call Transfer function in the voip unit. This is a "blind" transfer and the sequence of events is as follows: Callers A and B are having a conversation. Caller A wants to put B into contact with C. Caller A dials call transfer sequence. Caller A hears dial tone and dials number for caller C. Caller A gets disconnected while Caller B gets connected to caller C. A brief musical jingle is played for the caller on hold.
Transfer Sequence	any phone keypad character	The numbers and/or symbols that the caller must press on the phone keypad to initiate a call transfer. The call-transfer sequence can be 1 to 4 characters in length using any combination of digits or characters (* or #). The sequences for call transfer, call hold, and call waiting can be from 1 to 4 digits in length consisting of any combination of digits 1234567890*#.

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
Call Hold Enable	Y/N	Select to enable Call Hold function in voip unit. Call Hold allows one party to maintain an idle (non-talking) connection with another party while receiving another call (Call Waiting), while initiating another call (Call Transfer), or while performing some other call management function.
Hold Sequence	phone keypad characters	The numbers and/or symbols that the caller must press on the phone keypad to initiate a call hold. The call-hold sequence can be 1 to 4 characters in length using any combination of digits or characters (* or #).
Call Waiting Enable	Y/N	Select to enable Call Waiting function in voip unit.
Retrieve Sequence	phone keypad characters, two characters in length	The numbers and/or symbols that the caller must press on the phone keypad to initiate retrieval of a waiting call. The call-waiting retrieval sequence can be 1 to 4 characters in length using any combination of digits or characters (* or #). This is the phone keypad sequence that a user must press to retrieve a waiting call. Customize-able. Sequence should be distinct from sequence that might be used to retrieve a waiting call via the PBX or PSTN.

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
Call Name Identification Enable		Enables CNI function. Call Name Identification is not the same as Caller ID. When enabled on a given voip unit currently being controlled by the MultiVOIP GUI (the 'home voip'), Call Name Identification sends an identifier and status information to the administrator of the remote voip involved in the call. The feature operates on a channel-by-channel basis (each channel can have a separate identifier). If the home voip is originating the call, only the **Calling Party** field is applicable. If the home voip is receiving the call, then the **Alerting Party**, **Busy Party**, and **Connected Party** fields are the only applicable fields (and any or all of these could be enabled for a given voip channel). The status information confirms back to the originator that the callee (the home voip) is either busy, or ringing, or that the intended call has been completed and is currently connected. The identifier and status information are made available to the remote voip unit and appear in the **Caller ID** field of its **Statistics – Call Progress** screen. (This is how MultiVOIP units handle CNI messages; in other voip brands, H.450 may be implemented differently and then the message presentation may vary.)

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
Calling Party, Allowed Name Type (CNI)		If the 'home' voip unit is originating the call and **Calling Party** is selected, then the identifier (from the **Caller Id** field) will be sent to the remote voip unit being called. The **Caller Id** field gives the remote voip administrator a plain-language identifier of the party that is originating the call occurring on a specific channel. This field is applicable only when the 'home' voip unit is originating the call. **Example**. Suppose a voip system has offices in both Denver and Omaha. In the Omaha voip unit (the 'home' voip in this example), **Call Name Identification** has been enabled, **Calling Party** has been enabled as an **Allowed Name Type,** and "Omaha Sales Office Voipchannel 2" has been entered in the **Caller Id** field. When channel 2 of the Omaha voip is used to make a call to any other voip phone station (for example, the Denver office), the message "Calling Party - Omaha Sales Office Voipchannel 2" will appear in the "Caller Id" field of the **Statistics - Call Progress** screen of the Denver voip.

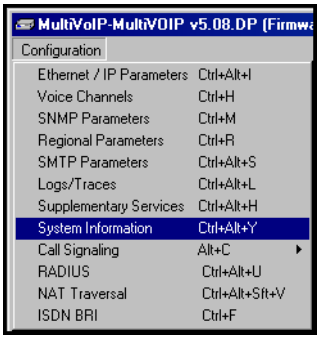
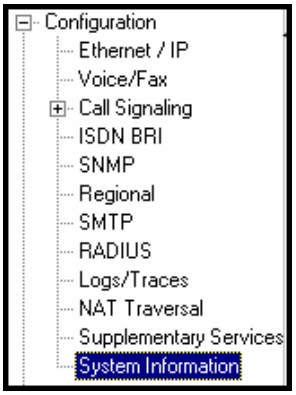
Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
Alerting Party, Allowed Name Type (CNI)		If the 'home' voip unit is receiving the call and **Alerting Party** is selected, then the identifier (from the **Caller Id** field) will tell the originating remote voip unit that the call is ringing. This field is applicable only when the 'home' voip unit is receiving the call. **Example**. Suppose a voip system has offices in both Denver and Omaha. In the Omaha voip unit (the 'home' voip unit in this example), **Call Name Identification** has been enabled, **Alerting Party** has been enabled as an **Allowed Name Type**, and "Omaha Sales Office Voipchannel 2" has been entered in the **Caller Id** field of the **Supplementary Services** screen. When channel 2 of the Omaha voip receives a call from any other voip phone station (for example, the Denver office), the message "Alerting Party - Omaha Sales Office Voipchannel 2" will be sent back and will appear in the **Caller Id** field of the **Statistics – Call Progress** screen of the Denver voip. This confirms to the Denver voip that the phone is ringing in Omaha.

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
Busy Party, Allowed Name Type (CNI)		If the 'home' voip unit is receiving a call directed toward an already engaged channel or phone station and **Busy Party** is selected, then the identifier (from the **Caller Id** field) will tell the originating remote voip unit that the channel or called party is busy. This field is applicable only when the 'home' voip unit is receiving the call. **Example**. Suppose a voip system has offices in both Denver and Omaha. In the Omaha voip unit (the 'home' voip unit in this example), **Call Name Identification** has been enabled, **Busy Party** has been enabled as an **Allowed Name Type**, and "Omaha Sales Office Voipchannel 2" has been entered in the **Caller Id** field of the **Supplementary Services** screen. When channel 2 of the Omaha voip is busy but still receives a call attempt from any other voip phone station (for example, the Denver office), the message "Busy Party - Omaha Sales Office Voipchannel 2" will be sent back and will appear in the **Caller Id** field of the **Statistics – Call Progress** screen of the Denver voip. This confirms to the Denver voip that the channel or phone station is busy in Omaha.

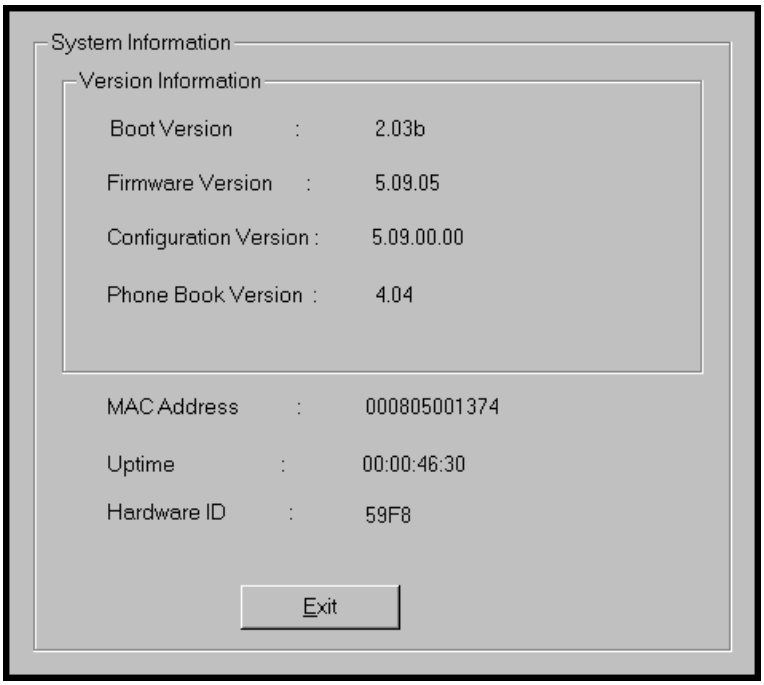
Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
Connected Party, Allowed Name Type (CNI)		If the 'home' voip unit is receiving a call and **Connected Party** is selected, then the identifier (from the **Caller Id** field) will tell the originating remote voip unit that the attempted call has been completed and the connection is made. This field is applicable only when the 'home' voip unit is receiving the call. **Example**. Suppose a voip system has offices in both Denver and Omaha. In the Omaha voip unit (the 'home' voip unit in this example), **Call Name Identification** has been enabled, **Connected Party** has been enabled as an **Allowed Name Type**, and "Omaha Sales Office Voipchannel 2" has been entered in the **Caller Id** field of the **Supplementary Services** screen. When channel 2 of the Omaha voip completes an attempted call from any other voip phone station (for example, the Denver office), the message "Connect Party - Omaha Sales Office Voipchannel 2" will be sent back and will appear in the **Caller Id** field of the **Statistics – Call Progress** screen of the Denver voip. This confirms to the Denver voip that the call has been completed to Omaha.

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
Caller ID		This is the identifier of a specific channel of the 'home' voip unit. The Caller Id field typically describes a person, office, or location, for example, "Harry Smith," or "Bursar's Office," or "Barnesville Factory."
Default	--	When this button is clicked, all Supplementary Service parameters are set to their default values.
Copy Channel	--	Copies the Supplementary Service attributes of one channel to another channel. Attributes can be copied to multiple channels or all channels at once.

19. View the **System Information** screen and set the updating interval (optional). This dialog box can be reached by pulldown menu, sidebar menu, or keyboard shortcut.

Accessing "System Information" Screen	
Pulldown	**Sidebar**
	
Shortcut	**Icon**
`Ctrl + Alt +Y`	--

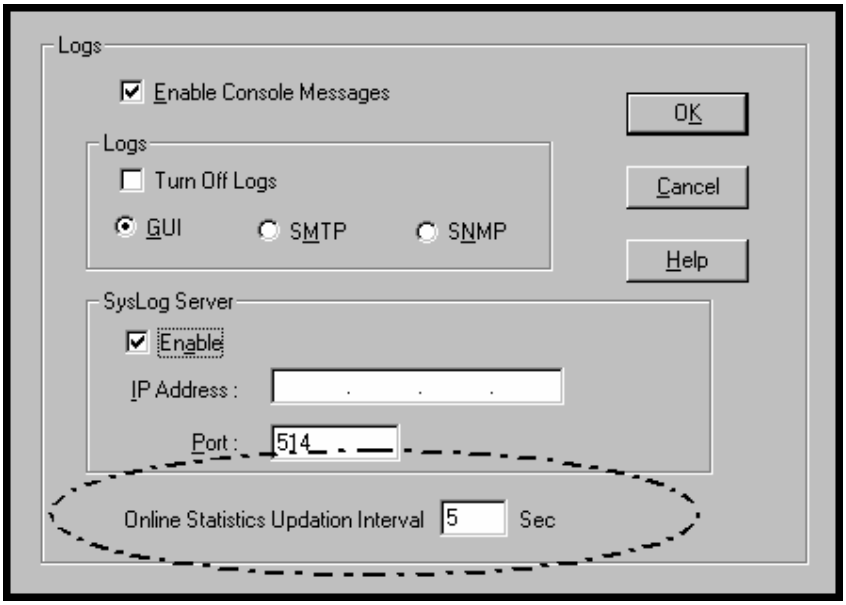
This screen presents vital system information at a glance. Its primary use is in troubleshooting.



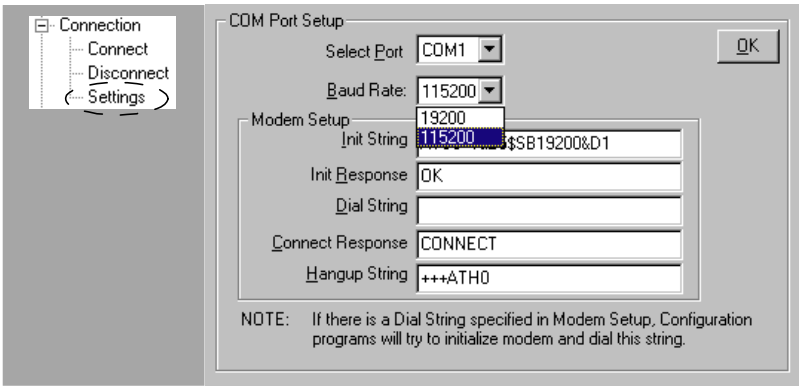
System Information Parameter Definitions		
Field Name	**Values**	**Description**
Boot Version	nn.nn	Indicates the version of the code that is used at the startup (booting) of the voip. The boot code version is independent of the software version.
Firmware Version	alpha-numeric	Indicates version of MultiVOIP firmware.

System Information Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Configur-ation Version	nn.nn.nn. nn alpha-numeric	Indicates version of MultiVOIP Configuration software (which includes screens for IP Parameters, SNMP Parameters, SMTP Parameters, Regional Parameters, etc.
Phone Book Version	numeric	Indicates the version of the inbound and outbound phonebook portion of the MultiVOIP software.
Mac Address	alpha-numeric	Denotes the number assigned as the voip unit's unique Ethernet address.
Up Time	days: hours: mm:ss	Indicates how long the voip has been running since its last booting.
Hardware ID	alpha-numeric	Indicates the version of the MultiVOIP unit's circuit board and components.

The frequency with which the System Information screen is updated is determined by a setting in the Logs screen.



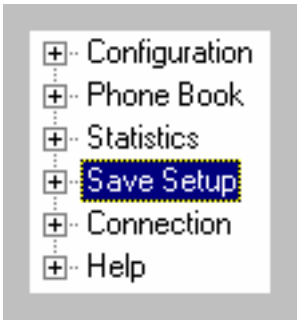
20. **Set Baud Rate**. The **Connection** option in the sidebar menu has a "Settings" item that includes the baud-rate setting for the COM port of the computer running the MultiVOIP software.



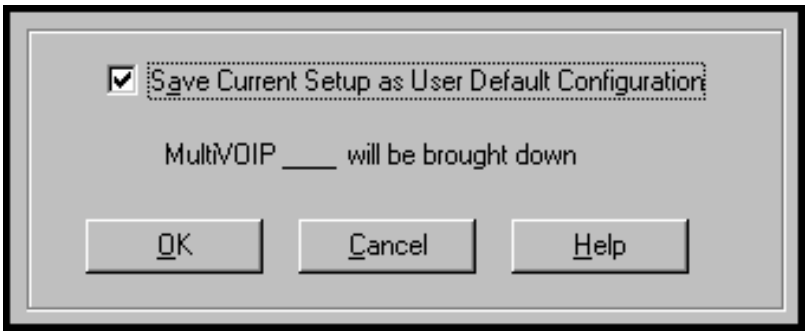
First, it is important to note that the default COM port established by the MultiVOIP program is COM1. ***Do not accept the default value until you have checked the COM port allocation on your PC***. To do this, check for COM port assignments in the system resource dialog box(es) of your Windows operating system. If COM1 is not available, you must change the COM port setting to COM2 or some other COM port that you have confirmed as being available on your PC.

The default baud rate is 115,200 bps.

21. **Saving the MultiVOIP Configuration**. When values have been set for all of the MultiVOIP's various operating parameters, click on **Save Setup** in the sidebar.



22. **Creating a User Default Configuration**. When a "Setup" (complete grouping of parameters) is being saved, you will be prepared about designating that setup as a "User Default" setup. A User Default setup may be useful as a baseline of site-specific values to which you can easily revert. Establishing a User Default Setup is optional.



Chapter 6: T1 Phonebook Configuration

(North American Telephony Standards)

T1 Versus E1 Telephony Environments

Phonebooks for Series II analog MultiVOIP units (MVP130, MVP130FXS, MVP210, MVP410, and MVP810) and ISDN-BRI MultiVOIPs (MVP-410ST/810ST) can be operated in either a North American telephony standards environment (potentially operating with T1 digital MultiVOIPs) or in a European telephony standards environment (potentially operating with E1 digital MultiVOIPs). The configuration of the phonebook is the same in either case. However, because the telephony environment is different in each case and the examples used here must reflect those differences, we have separate chapters for phonebook configuration in North American (T1) environments (Chapter 6) and for that in European (E1) environments (Chapter 7). Consult the chapter that best fits the needs of your voip system. Phonebook configuration examples shown for Series II analog MultiVOIPs are fully applicable to ISDN-BRI MultiVOIPs except where the telephony interface is concerned.

Configuring T1 (NAM) Telephony MultiVOIP Phonebooks

When a VoIP serves a PBX system, it's important that the operation of the VoIP be transparent to the telephone end user. That is, the VoIP should not entail the dialing of extra digits to reach users elsewhere on the network that the VoIP serves. On the contrary, VOIP service more commonly reduces dialed digits by allowing users (served by PBXs in facilities in distant cities) to dial their co-workers with 3-, 4-, or 5-digit extensions as if they were in the same facility.

Furthermore, the setup of the VoIP generally should allow users to make calls on a non-toll basis to any numbers accessible without toll by users at all other locations on the VoIP system. Consider, for example, a company with VOIP-equipped offices in New York, Miami, and Los Angeles, each served by its own PBX. When the VOIP phone books are set correctly, personnel in the Miami office should be able to make calls without toll not only to the company's offices in New York and Los Angeles, but also to any number that's local in those two cities.

To achieve transparency of the VoIP telephony system and to give full access to all types of non-toll calls made possible by the VOIP system, the VoIP administrator must properly configure the "Outbound" and "Inbound" phone-books of each VoIP in the system.

The "Outbound" phonebook for a particular VoIP unit describes the dialing sequences required for a call to originate locally (typically in a PBX in a particular facility) and reach any of its possible destinations at remote VoIP sites, including non-toll calls completed in the PSTN at the remote site.

The "Inbound" phonebook for a particular VoIP unit describes the dialing sequences required for a call to originate remotely from any other VOIP sites in the system, and to terminate on that particular VOIP.

Briefly stated, *the MultiVOIP's Outbound phonebook lists the phone stations it can call; its Inbound phonebook describes the dialing sequences that can be used to call that MultiVOIP and how those calls will be directed.* (Of course, the phone numbers are not literally "listed" individually, but are, instead, described by rule.)

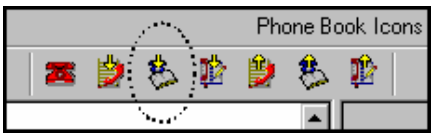
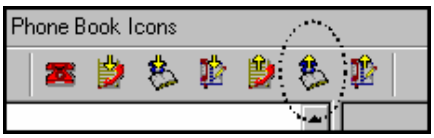
Consider two types of calls in the three-city system described above: (1) calls originating from the Miami office and terminating in the New York (Manhattan) office, and (2) calls originating from the Miami office and terminating in New York City but off the company's premises in an adjacent area code, an area code different than the company's office but still a local call from that office (e.g., Staten Island).

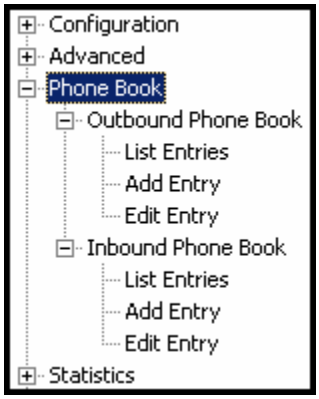
The first type of call requires an entry in the Outbound PhoneBook of the Miami VOIP and a coordinated entry in the Inbound phonebook of the New York VOIP. These entries would allow the Miami caller to dial the New York office as if its phones were extensions on the Miami PBX.

The second type of call similarly requires an entry in the Outbound PhoneBook of the Miami VOIP and a coordinated entry in the Inbound Phonebook of the New York VOIP. However, these entries will be longer and more complicated. Any Miami call to New York City local numbers will be sent through the VOIP system rather than through the regular toll public phone system (PSTN). But the phonebook entries can be arranged so that the VOIP system is transparent to the Miami user, such that even though that Miami user dials the New York City local number just as they would through the public phone system, that call will still be completed through the VOIP system.

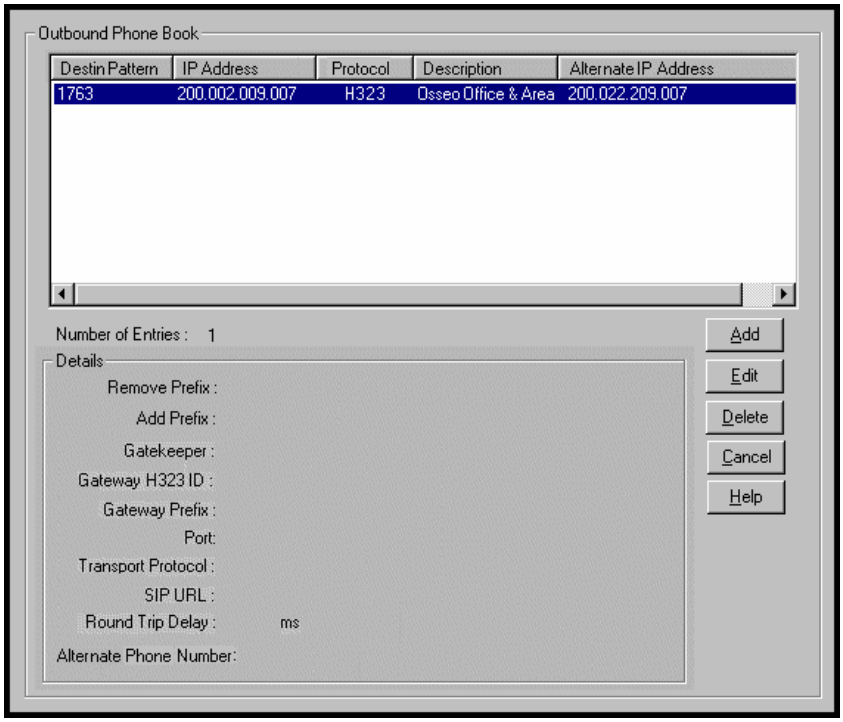
This PhoneBook Configuration procedure is brief, but it is followed by an example case. For many people, the example case may be easier to grasp than the procedure steps. Configuration is not difficult, but all phone number sequences and other information must be entered exactly; otherwise connections will not be made.

Phonebook configuration screens can be accessed using icons or the sidebar menu.

Phonebook Icons	Description
	Phonebook Configuration
	Inbound Phonebook Entries List
	Add Inbound Phonebook Entry
	Edit selected Inbound Phonebook Entry
	Outbound Phonebook Entries List
	Add Outbound Phonebook Entry
	Edit selected Outbound Phonebook Entry

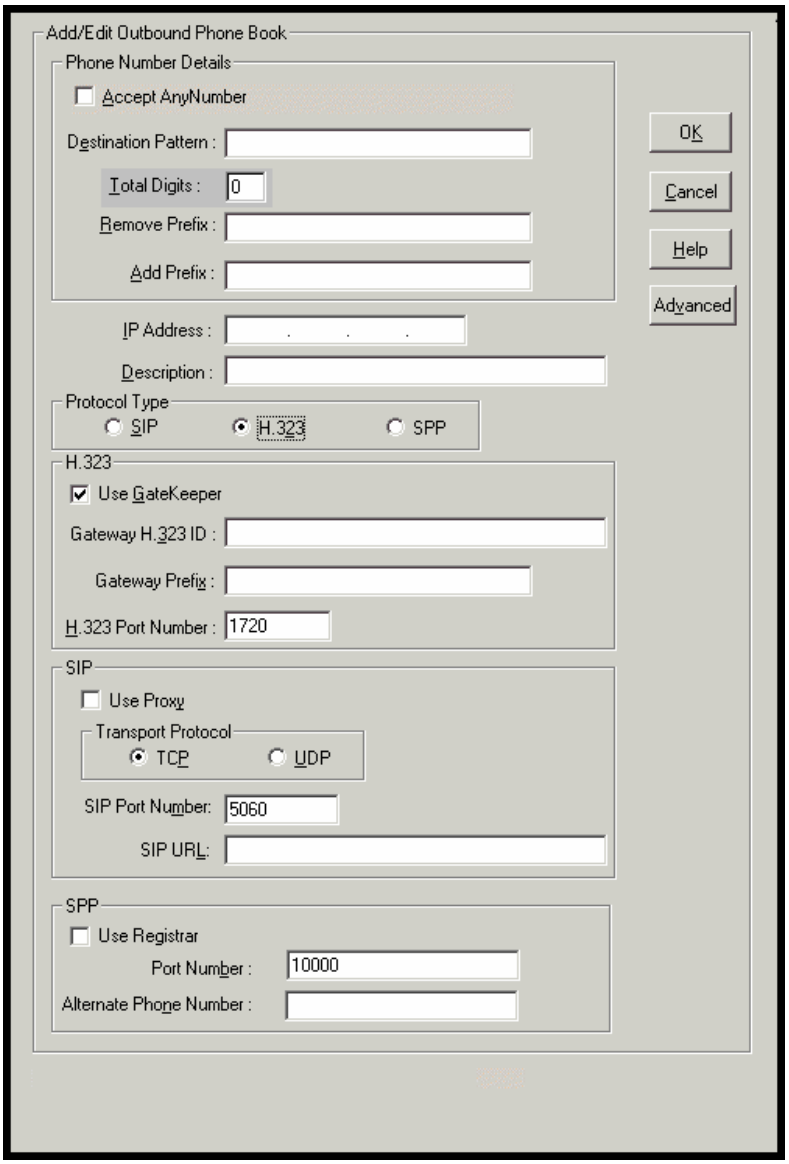
Phonebook Pulldown Menu	
Inbound Phonebook Shortcut	**Outbound Phonebook Shortcut**
Alt + I	Alt + O
Phonebook Sidebar Menu	
	

1. Select **Outbound Phone Book/List Entries**.
   Fields in the "Details" section will differ depending on the protocol (H.323, SIP, or SPP) of the selected list entry to which the details pertain.



   Click **Add**.

2. The **Add/Edit Outbound PhoneBook** screen appears.



Enter Outbound PhoneBook data for your MultiVOIP unit. Note that the Advanced button gives access to the Alternate IP Routing feature, if needed. Alternate IP Routing can be implemented in a secondary screen (as described after the primary screen field definitions below).

The fields of the **Add/Edit Outbound Phone Book** screen are described in the table below.

Add/Edit Outbound Phone Book: Field Definitions		
Field Name	**Values**	**Description**
Accept Any Number	Y/N	When checked, "Any Number" appears as the value in the Destination Pattern field. The **Accept Any Number** feature works differently depending on whether or not an external routing device is used (Gatekeeper for H323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). **When no external routing device is used**. If **Accept Any Number** is selected, calls to phone numbers not matching a listed Destination Pattern will be directed to the **IP Address** in the Add/Edit Outbound Phone Book screen. "Any Number" can be used in addition to one or more Destination Patterns. **When external routing device is used**. If Any Number is selected, calls to phone numbers not matching a listed Destination Pattern will be directed to the external routing device used (Gatekeeper for H323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). The IP Address of the external routing device must be set in the Phone Book Configuration screen.

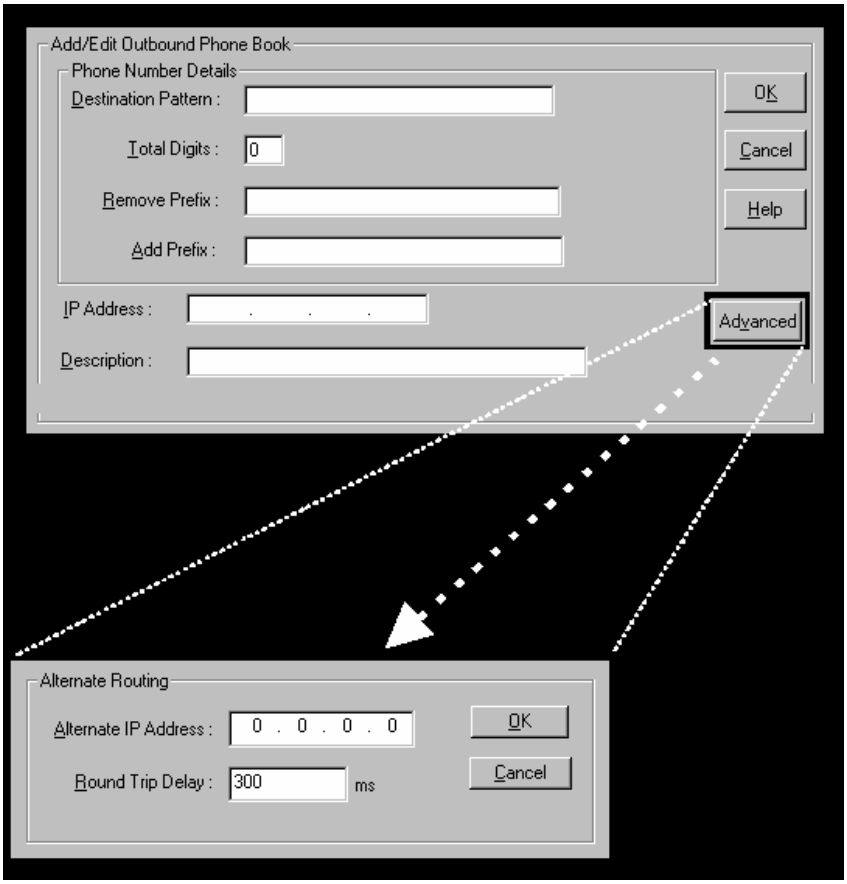
Add/Edit Outbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Destination Pattern	prefixes, area codes, exchanges, line numbers, extensions	Defines the beginning of dialing sequences for calls that will be connected to another VOIP in the system. Numbers beginning with these sequences are diverted from the PTSN and carried on Internet or other IP network.
Total Digits	as needed	*This field currently disabled.* number of digits the phone user must dial to reach specified destination.
Remove Prefix	dialed digits	portion of dialed number to be removed before completing call to destination
Add Prefix	dialed digits	digits to be added before completing call to destination
IP Address	n.n.n.n for n = 0-255	the IP address to which the call will be directed if it begins with the destination pattern given
Description	alpha-numeric	Describes the facility or geographical location at which the call will be completed.
Protocol Type	SIP or H.323 or SPP	Indicates protocol to be used in outbound transmission. Single Port Protocol (SPP) is a non-standard protocol designed by Multi-Tech.

Add/Edit Outbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
H.323 fields		
Use Gatekeepr	Y/N	Indicates whether or not gatekeeper is used.
Gateway H.323 ID	alpha-numeric	The H.323 ID assigned to the destination MultiVOIP. Only valid if "Use Gatekeeper" is enabled for this entry.
Gateway Prefix	numeric	This number becomes registered with the GateKeeper. Call requests sent to the gatekeeper and preceded by this prefix will be routed to the VOIP gateway.
H.323 Port Number	1720	This parameter pertains to Q.931, which is the H.323 call signaling protocol for setup and termination of calls (aka ITU-T Recommendation I.451). H.323 employs only one "well-known" port (1720) for Q.931 signaling. If Q.931 message-oriented signaling protocol is used, 1720 must be chosen as the H.323 Port Number.

Add/Edit Outbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
SIP Fields		
Use Proxy	Y/N	Select if proxy server is used.
Transport Protocol	TCP or UDP	Voip administrator must choose between UDP and TCP transmission protocols. UDP is a high-speed, low-overhead connectionless protocol where data is transmitted without acknowledgment, guaranteed delivery, or guaranteed packet sequence integrity. TCP is slower connection-oriented protocol with greater overhead, but having acknowledgment and guarantees delivery and packet sequence integrity.
SIP Port Number	5060 or other *See RFC 3087 ("Control of Service Context using SIP Request-URI," by the Network Working Group).	The SIP Port Number is a UDP logical port number. The voip will "listen" for SIP messages at this logical port. If SIP is used, 5060 is the default, standard, or "well known" port number to be used. If 5060 is not used, then the port number used is that specified in the SIP Request URI (Universal Resource Identifier).
SIP URL	*sip.userphone @ hostserver*, where "userphone" is the telephone number and "hostserver"is the domain name or an address on the network	Looking similar to an email address, a **SIP URL** identifies a user's address. In SIP communications, each caller or callee is identified by a SIP url: sip:user_name@host_name. The format of a sip url is very similar to an email address, except that the "sip:" prefix is used.

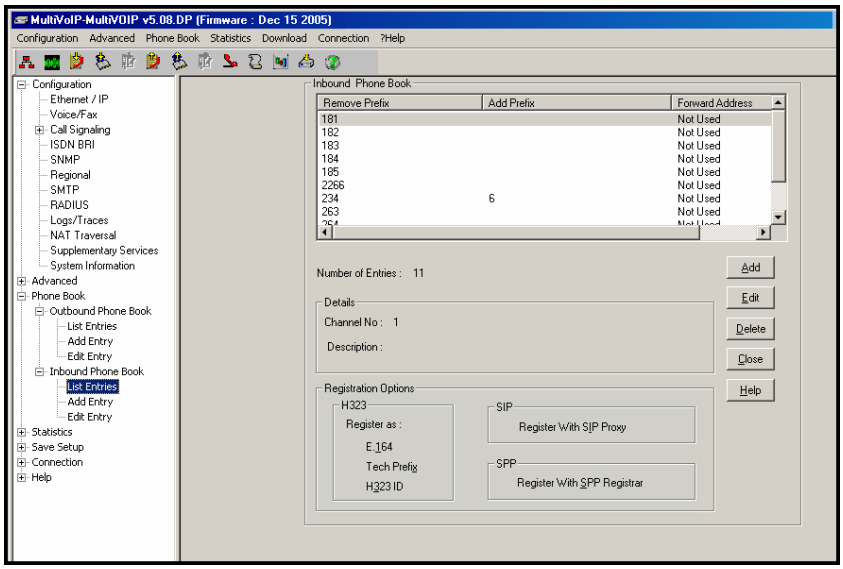
Add/Edit Outbound Phone Book: Field Def'ns (cont'd)		
Field Name	**Values**	**Description**
SPP Fields		
Use Registrar	**Values:** Y/N **Description:** Select this checkbox to use registrar when voip system is operating in the "Registrar/Client" SPP mode. In this mode, one voip (the registrar, as set in Phonebook Configuration screen) has a static IP address and all other voips (clients) point to the registar's IP address as functionally their own. However, if your voip system overall is operating in "Registrar/Client" mode but you want to make an exception and use Direct mode for the destination pattern of this particular Add/Edit Phonebook entry, leave this checkbox unselected. Leave this checkbox unselected if your overall voip system is operating in the "Direct" SPP mode. In this mode, all voips in system are peers and each has its own static IP address.	
Port Number	**Values:** numeric **Description:** When operating in "Registrar/Client" mode, this is the port by which the gateway receives all SPP data and control messages from the registrar gateway. (This ability to receive all data and messages via one port allows the voip to operate behind a firewall with only one port open.) When operating in "Direct" mode, this is the Port by which peer voips receive data and messages.	
Alternate Phone Number	numeric	Phone number associated with alternate IP routing.
Remote Device is [legacy voip]	Y/N	When checked, this MultiVOIP can operate with 'first-generation' MultiVOIP units in the same IP network. These include MVP-110/120/200/400/800.
Advanced button	**Values:** N/A **Description:** Gives access to secondary screen where an Alternate IP Route can be specified for backup or redundancy of signal paths. See discussion on next page. For SIP & H.323 operation only.	

Clicking on the **Advanced** button brings up the **Alternate Routing** secondary screen. This feature provides an alternate path for calls if the primary IP network cannot carry the traffic. Often in cases of failure, call traffic is temporarily diverted into the PSTN. However, this feature could also be used to divert traffic to a redundant (backup) unit in case one voip unit fails. The user must specify the IP address of the alternate route for each destination pattern entry in the Outbound Phonebook.

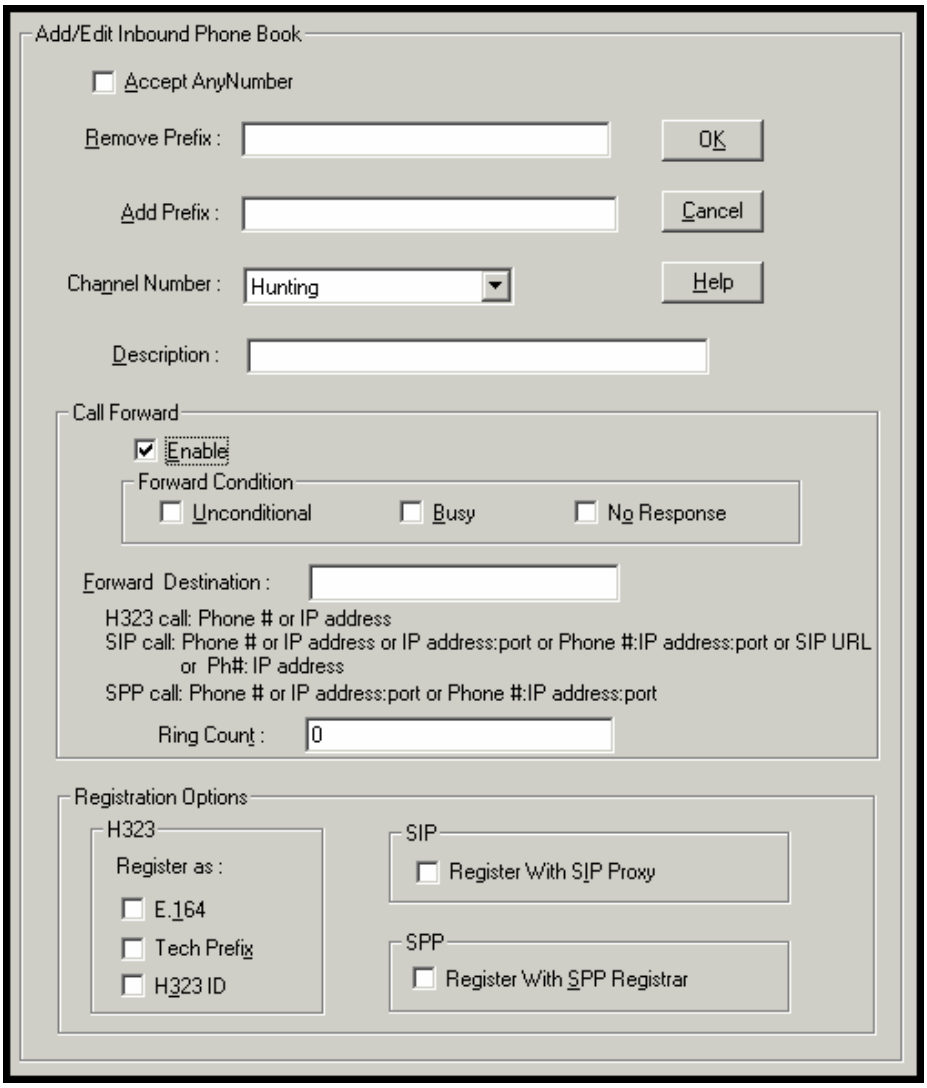


Alternate Routing Field Definitions		
Field Name	**Values**	**Description**
Alternate IP Address	n.n.n.n where n= 0-255	Alternate destination for outbound data traffic in case of excessive delay in data transmission.
Round Trip Delay	milliseconds	The Round Trip Delay is the criterion for judging when a data pathway is considered blocked. When the delay exceeds the threshold specified here, the data stream will be diverted to the alternate destination specified as the Alternate IP Address.

3. Select **Inbound PhoneBook | List Entries**.



4. The **Add/Edit Inbound PhoneBook** screen appears.



Enter Inbound PhoneBook data for your MultiVOIP. The fields of the Add/Edit Inbound PhoneBook screen are described in the table below.

Add/Edit Inbound Phone Book: Field Definitions		
Field Name	**Values**	**Description**
Accept Any Number	**Values**: Y/N **Description:** When checked, "Any Number" appears as the value in the Remove Prefix field. The **Any Number** feature of the Inbound Phone Book does not work when an external routing device is used (Gatekeeper for H323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). **When no external routing device is used**. If **Any Number** is selected, calls received from phone numbers not matching a listed Prefix (shown in the Remove Prefix column of the Inbound Phone Book) will be admitted into the voip on the channel listed in the **Channel Number** field. "Any Number" can be used in addition to one or more Prefixes.	
Remove Prefix	dialed digits	portion of dialed number to be removed before completing call to destination (often a local PBX)
Add Prefix	dialed digits	digits to be added before completing call to destination (often a local PBX)
Channel Number	1-24, or "Hunting"	T1 channel number to which the call will be assigned as it enters the local telephony equipment (often a local PBX). "Hunting" directs the call to any available channel.
Description	--	Describes the facility or geographical location at which the call originated.
Call Forward Parameters		
Enable	Y/N	Click the check-box to enable the call-forwarding feature.

Add/Edit Inbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Call Forward Parameters		
Forward Condition	Uncondit.; Busy No Resp.	**Unconditional**. When selected, all calls received will be forwarded. **Busy**. When selected, calls will be forwarded when station is busy. **No Response**. When selected, calls will be forwarded if called party does not answer after a specified number of rings, as specified in **Ring Count** field. Forwarding can be conditioned on both "Busy" and "No Response."
Forward Destination IP address, phone number, port number, etc.	Phone number or IP address to which calls will be directed. For H.323 calls, the Forward Destination can be either a Phone Number or an IP Address. For SIP calls, the Forward Destination can be one of the following: (a) phone number, (b) IP address, (c) IP address: port number, (d) phone number:IP addr: port number, (e) SIP URL, or (f) phone #: IP address. For SPP calls, the Forward Destination can be one of the following: (a) phone number, (b) IP address: port, or (c) phone number: IP address: port.	

Add/Edit Inbound Phone Book: Field Definitions (cont'd)	
Field Name	**Values and Description**
Ring Count	0, 1, 2, 3, etc. When "No Response" is condition for forwarding calls, this determines how many unanswered rings are needed to trigger the forwarding.
Registration Option Parameters	In an H.323 voip system, gateways can register with the system using one of these identifiers: (a) an E.164 identifier, (b) a Tech Prefix identifier, or (c) an H.323 ID identifier. In a SIP voip system, gateways can register with the SIP Proxy. In an SPP voip system, gateways can register with the SPP Registrar voip unit.

5. When your Outbound and Inbound PhoneBook entries are completed, click on **Save Setup** in the sidebar menu to save your configuration.

You can change your configuration at any time as needed for your system.

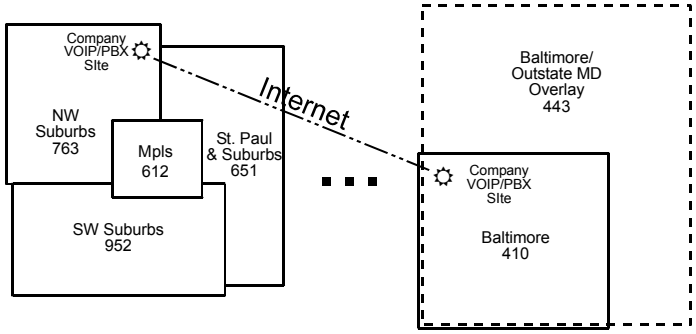
Remember that the initial MultiVOIP setup must be done locally using the MultiVOIP Windows GUI or remotely using the MultiVOIP web GUI (contacting the remote MultiVOIP initially at its default IP address of 192.168.3.143). After the initial configuration is complete, all of the MultiVOIP units in the VOIP system can be configured, re-configured, and updated from one location using the MultiVOIP web GUI software program or the MultiVOIP program (in conjunction with the built-in modem).

T1 Phonebook Examples

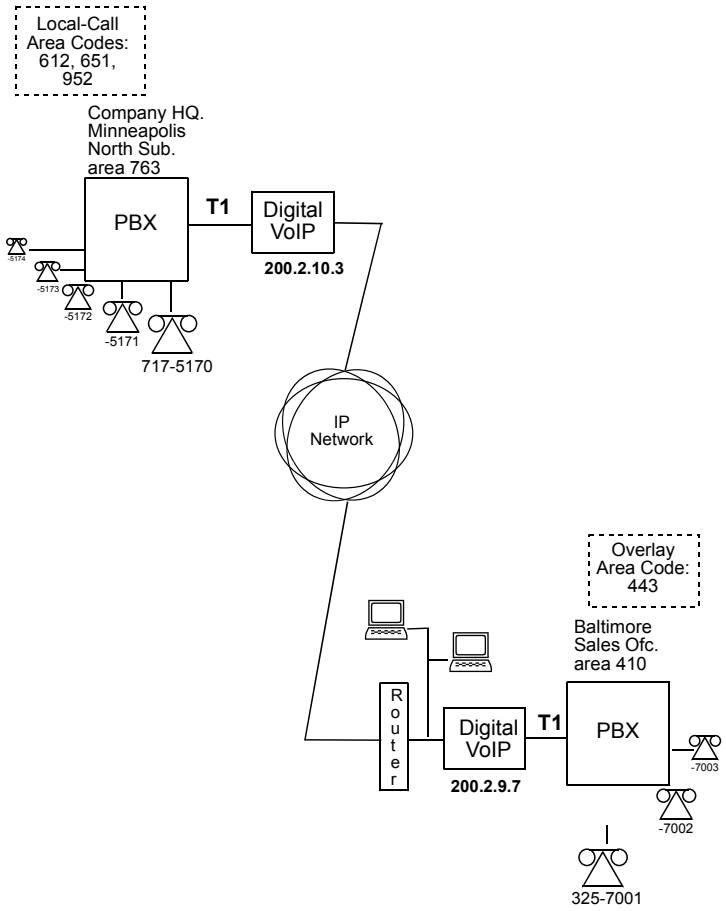
The following example demonstrates how Outbound and Inbound PhoneBook entries work in a situation of multiple area codes. Consider a company with offices in Minneapolis and Baltimore.

3 Sites, All-T1 Example

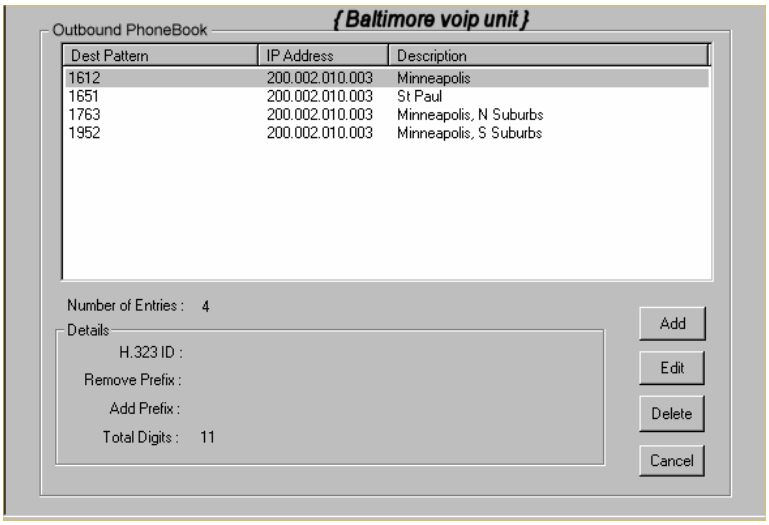
Notice first the area code situation in those two cities: Minneapolis's local calling area consists of multiple adjacent area codes; Baltimore's local calling area consists of a base area code plus an overlay area code.



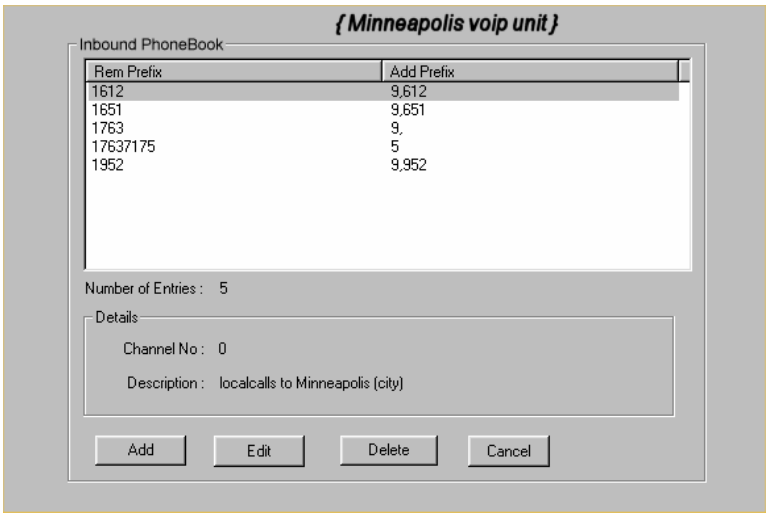
An outline of the equipment setup in both offices is shown below.



The screen below shows Outbound PhoneBook entries for the VOIP located in the company's Baltimore facility.



The entries in the Minneapolis VOIP's Inbound PhoneBook match the Outbound PhoneBook entries of the Baltimore VOIP, as shown below.



To call the Minneapolis/St. Paul area, a Baltimore employee must dial eleven digits. (In this case, we are assuming that the Baltimore PBX does not require an "8" or "9" to seize an outside phone line.)

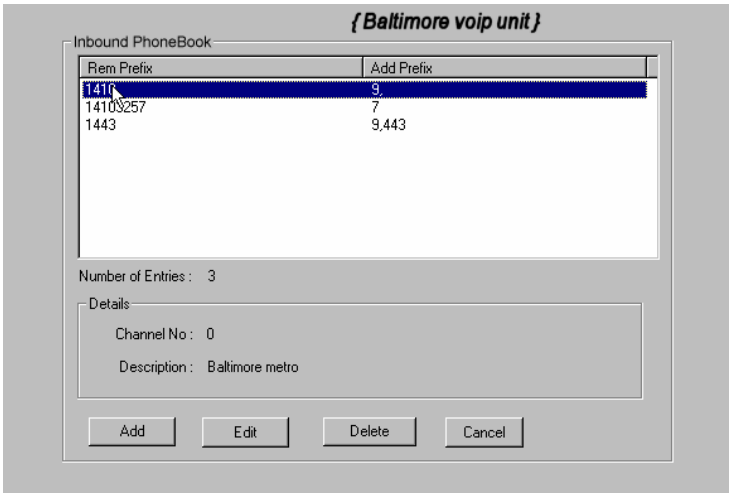
If a Baltimore employee dials any phone number in the 612 area code, the call will automatically be handled by the company's voip system. Upon receiving such a call, the Minneapolis voip will remove the digits "1612". But before the suburban-Minneapolis voip can complete the call to the PSTN of the Minneapolis local calling area, it must dial "9" (to get an outside line from the PBX) and then a comma (which denotes a pause to get a PSTN dial tone) and then the 10-digit phone number which includes the area code (612 for the city of Minneapolis; which is different than the area code of the suburb where the PBX is actually located -- 763).

A similar sequence of events occurs when the Baltimore employee calls number in the 651 and 952 area codes because number in both of these area codes are local calls in the Minneapolis/St. Paul area.

The simplest case is a cal from Baltimore to a phone within the Minneapolis/St. Paul area code where the company's voip and PBX are located, namely 763. In that case, that local voip removes 1763 and dials 9 to direct the call to its local 7-digit PSTN.

Finally, consider the longest entry in the Minneapolis Inbound Phonebook, "17637175. Note that the main phone number of the Minneapolis PBX is 763-717-5170. The destination pattern 17637175 means that all calls to Minneapolis employees will stay within the suburban Minneapolis PBX and will not reach or be carried on the local PSTN.

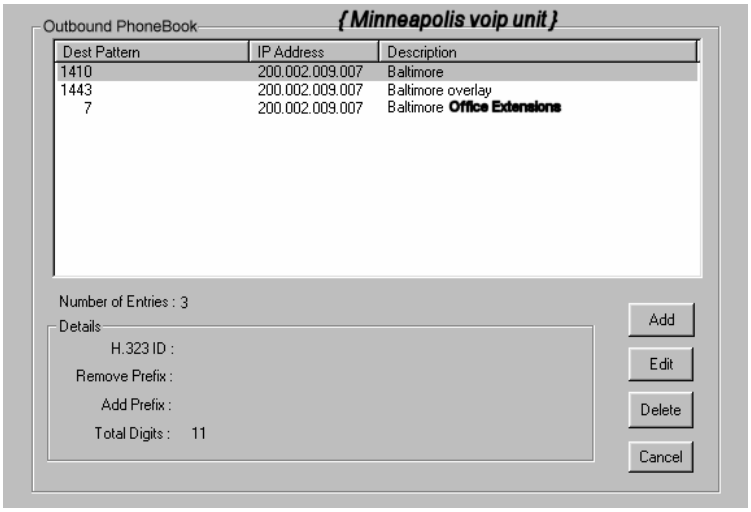
Similarly, the Inbound PhoneBook for the Baltimore VOIP (shown first below) generally matches the Outbound PhoneBook of the Minneapolis VOIP (shown second below).



Notice the extended prefix to be removed: 14103257. This entry allows Minneapolis users to contact Baltimore co-workers as though they were in the Minneapolis facility, using numbers in the range 7000 to 7999.

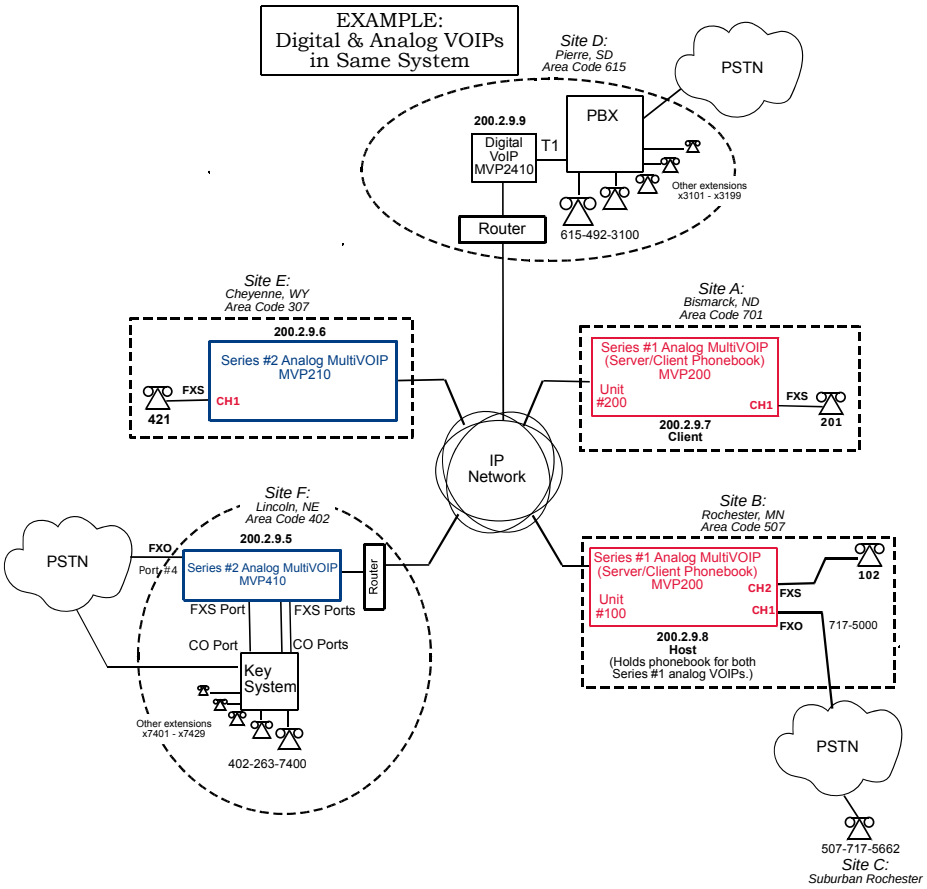
Note also that a comma (as in the entry 9,443) denotes a delay in dialing. A one-second delay is commonly used to allow a second dial tone to be generated for calls going outside of the facility's PBX system.

The Outbound PhoneBook for the Minneapolis VOIP is shown below. The third destination pattern, "7" facilitates reception of co-worker calls using local-appearing-extensions only. In this case, the "Add Prefix" field value for this phonebook entry would be "1410325" .



Configuring Mixed Digital/Analog VOIP Systems

Analog MultiVOIP units, like the MVP-210/410/810 are compatible with digital MultiVOIP units like the MVP2410. In many cases, digital and analog VOIP units will appear in the same telephony/IP system. In addition to MVP-210/410/810 MultiVOIP units (Series II units), legacy analog VOIP units (Series I units made by MultiTech) may be included in the system, as well. When legacy VOIP units are included, the VOIP administrator must handle two styles of phonebooks in the same VOIP network. The diagram below shows a small-scale system of this kind: one digital VOIP (the MVP2410) operates with two Series II analog VOIPs (an MVP210 and an MVP410), and two Series I legacy VOIPs (two MVP200 units).



The Series I analog VOIP phone book resides in the “Host” VOIP unit at Site B. It applies to both of the Series I analog VOIP units.

Each of the Series II analog MultiVOIPs (the MVP210 and the MVP410) requires its own inbound and outbound phonebooks. The MVP2410 digital MultiVOIP requires its own inbound and outbound phonebooks, as well.

These seven phone books are shown below.

Phone Book for Series I Analog VOIP Host Unit (Site B)			
VOIP Dir # -OR- Destination Pattern	IP Address	Channel	Comments
102	200.2.9.8	2	Site B, FXS channel.
101	200.2.9.8	1	Site B, FXO channel.
421	200.2.9.6	0	Site E FXS channel.
201	200.2.9.7	1	Site A, FXS channel.
1615 xxx xxxx	200.2.9.9	0 (Note 2.)	Gives remote voip users access to local PSTN of Site D (Pierre, SD, area code 615).
3xxx (Note 1.)	200.2.9.9	0	Allows remote voip users to call all PBX extensions at Site D (Pierre, SD) using only four digits.
1402	200.2.9.5	0	Gives remote voip users access to local PSTN of Site F (Lincoln, NE; area code 402).
140226374 (Note 1) (Note 3)	200.2.9.5	0	Gives remote voip users access to key phone system extensions at Site F (Lincoln).

Note 1. The "x" is a wildcard character.

Note 2. By specifying "Channel 0," we instruct the MVP2400/2410 to choose any available data channel to carry the call.

Note 3. Note that Site F key system has only 30 extensions (x7400-7429). This destination pattern (140226374) actually directs calls to 402-263-74**30** through 402-263-74**99** into the key system, as well. This means that such calls, which belong on the PSTN, cannot be completed. In some cases, this might be inconsequential because an entire exchange (fully used or not) might have been reserved for the company or it might be unnecessary to reach those numbers. However, to specify only the 30 lines actually used by the key system, the destination pattern 140226374 would have to be replaced by three other destination patterns, namely 1402263740, 1402263741, and 1402263742. In this way, calls to 402-263-74**30** through 402-263-74**99** would be properly directed to the PSTN. In the Site D outbound phonebook, the 30 lines are defined exactly, that is, without making any adjacent phone numbers unreachable through the voip system.

Outbound Phone Book for MVP2410 Digital VOIP (Site D)				
Destin. Pattern	Remove Prefix	**Add Prefix**	**IP Address**	**Comment**
201			200.2.9.7	To originate calls to Site A (Bismarck).
1507	1507	101# Note 3.	200.2.9.8	To originate calls to Rochester local PSTN using the FXO channel (channel #1) of the Site B VOIP.
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP.
421			200.2.9.6	Calls to Site E (Cheyenne).
1402			200.2.9.5	Calls to Lincoln area local PSTN (via FXO channel, CH4, of the Site F VOIP).
1402 263 740			200.2.9.5	Calls to extensions (thirty) of key system at Site F (Lincoln). Human operator or auto-attendant is needed to complete these calls.
1402 263 741			200.2.9.5	
1402 263 742			200.2.9.5	
Note 3. The pound sign ("#") is a delimiter separating the VOIP number from the standard telephony phone number.				

Inbound Phonebook for MVP2410 Digital VOIP (Site D)			
Remove Prefix	**Add Prefix**	**Channel Number**	**Comment**
1615	9, Note 4. Note 5.	0	Allows phone users at remote voip sites to call non-toll numbers within the Site D area code (615; Pierre, SD) over the VOIP network.
1615 49231	31	0	Allows voip calls directly to employees at Site D (at extensions x3101 to x3199).

Note 4. "9" gives PBX station users access to outside line.

Note 5. The comma represents a one-second pause, the time required for the user to receive a dial tone on the outside line (PSTN). The comma is only allowed in the Inbound phonebook.

Outbound Phone Book for MVP410 Analog VOIP (Site F)				
Destin. Pattern	Remove Prefix	**Add Prefix**	**IP Address**	**Comment**
201			200.2.9.7	To originate calls to Site A (Bismarck).
1507	1507	101# Note 3.	200.2.9.8	To originate calls to any PSTN phone in Rochester area using the FXO channel (channel #1) of the Site B VOIP.
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP (Rochester).
421			200.2.9.6	Calls to Site E (Cheyenne).
1615			200.2.9.9	Calls to Pierre area PSTN via Site D PBX.
31		1615 492	200.2.9.9	Calls to Pierre PBX extensions with four digits.
Note 3. The pound sign ("#") is a delimiter separating the VOIP number from the standard telephony phone number.				

Inbound Phonebook for MVP410 Analog VOIP (Site F)			
Remove Prefix	**Add Prefix**	**Channel Number**	**Comment**
1402		4	Access to Lincoln local PSTN by users at remote VOIP locations via FXO port at Site F.
1402 263740	740	0	Gives remote voip users access to extension of key phone system at Site F (Lincoln). Because call is completed at key system, abbreviated dialing (4 digits) is not workable. Human operator or auto-attendant is needed to complete these calls.
1402 263741	741	0	
1402 263742	742	0	

Outbound Phone Book for MVP210 Analog VOIP (Site E)				
Destin. Pattern	Remove Prefix	**Add Prefix**	**IP Address**	**Comment**
201			200.2.9.7	To originate calls to Site A.
1507	1507	101# Note 3.	200.2.9.8	To originate calls to any PSTN phone in Rochester area using the FXO channel (channel #1) of the Site B VOIP.
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP.
1402			200.2.9.5	Calls to Lincoln area PSTN (via FXO channel, CH4, of the Site F VOIP).
7		1402 263	200.2.9.5	Calls to Lincoln key extensions with four digits.
1615			200.2.9.9	Calls to Pierre area PSTN via Site D PBX.
31		1615 492	200.2.9.9	Calls to Pierre PBX extensions with four digits.
Note 3. The pound sign ("#") is a delimiter separating the VOIP number from the standard telephony phone number.				

Inbound Phonebook for MVP210 Analog VOIP (Site E)			
Remove Prefix	**Add Prefix**	**Channel Number**	**Comment**
421		1	

Call Completion Summaries

Site A calling Site C, Method 1

1. Dial 101.
2. Hear dial tone from Site B.
3. Dial 7175662.
4. Await completion. Talk.

Site A calling Site C, Method 2

1. Dial 101#7175662
2. Await completion. Talk.

> Note: Some analog VOIP gateways will allow completion by Method 2. Others will not.

Site C calling Site A

1. Dial 7175000.
2. Hear dial tone from Site B VOIP.
3. Dial 201.
4. Await completion. Talk.

Site D calling Site C

1. Dial 9,15077175662.
2. "9" gets outside line. On some PBXs, an "8" may be used to direct calls to the VOIP, while "9" directs calls to the PSTN. However, some PBX units can be programmed to identify the destination patterns of all calls to be directed to the VOIP.
3. PBX at Site D is programmed to divert all calls made to the 507 area code and exchange 717 into the VOIP network. (It would also be possible to divert all calls to all phones in area code 507 into the VOIP network, but it may not be desirable to do so.)
4. The MVP2410 removes the prefix "1507" and adds the prefix "101#" for compatibility with the analog MultiVOIP's phonebook scheme. The "#" is a delimiter separating the analog VOIP's phone number from the digits that the analog VOIP must dial onto its local PSTN to complete the call. The digits "101#7175662" are forwarded to the Site B analog VOIP.
5. The call passes through the IP network (in this case, the Internet).
6. The call arrives at the Site B VOIP. This analog VOIP receives this dialing string from the MVP2410: 101#7175662. The analog VOIP, seeing the "101" prefix, uses its own channel #1 (an FXO port) to connect the call to the PSTN. Then the analog VOIP dials its local phone number 7175662 to complete the call.

Site D calling Site F

A voip call from Pierre PBX to extension 7424 on the key telephone system in Lincoln, Nebraska.

A. The required entry in the Pierre Outbound Phonebook to facilitate origination of the call, would be 1402263742. The call would be directed to the Lincoln voip's IP address, 200.2.9.5.
(Generally on such a call, the caller would have to dial an initial "9." But typically the PBX would not pass the initial "9" to the voip. If the PBX *did* pass along that "9" however, its removal would have to be specified in the local Outbound Phonebook.)

B. The corresponding entry in the Lincoln Inbound Phonebook to facilitate completion of the call would be

1402263742	for calls within the office at Lincoln
1402	for calls to the Lincoln local calling area (PSTN).

Call Event Sequence

1. Caller at Pierre dials 914022637424.
2. Pierre PBX removes "9" and passes 14022637424 to voip.
3. Pierre voip passes remaining string, 14022637424 on to the Lincoln voip
   at IP address 200.2.9.5.
4. The dialed string matches an inbound phonebook entry at the Lincoln voip, namely 1402263742.
5. The Lincoln voip rings one of the three FXS ports connected to the Lincoln
   key phone system.
6. The call will be routed to extension 7424 either by a human receptionist/
   operator or to an auto-attendant (which allows the caller to specify the
   extension to which they wish to be connected).

Site F calling Site D

A voip call from a Lincoln key extension to extension 3117 on the PBX in Pierre, South Dakota.

A. The required entry in the Lincoln Outbound Phonebook to facilitate origination of the call, would be "31". The string "1615492" would have to be added as a prefix. The call would be directed to the Pierre voip's IP address, 200.2.9.9.

B. The corresponding entry in the Pierre Inbound Phonebook to facilitate completion of the call would be 1615492.

1. Caller at Lincoln picks up phone receiver, presses button on key phone set. This button has been assigned to a particular voip channel (any one of the three FXS ports).
2. The caller at Lincoln hears dial tone from the Lincoln voip.
3. The caller at Lincoln dials 3117.
4. The Lincoln voip adds the prefix 1615492 and sends the entire dialing string, 16154923117, to the Pierre voip at IP address 200.2.9.9.
5. The Pierre voip matches the called digits 16154923117 to its Inbound Phonebook entry "1615492" .
6. The Pierre PBX dials extension 3117 in the office at Pierre.

Variations in PBX Characteristics

The exact dialing strings needed in the Outbound and Inbound Phonebooks of the MVP2410 will depend on the capabilities of the PBX. Some PBXs require trunk access codes (like an "8" or "9" to access an outside line or to access the VOIP network). Other PBXs can automatically distinguish between intra-PBX calls, PSTN calls, and VOIP calls.

Some PBX units can also insert digits automatically when they receive certain dialing strings from a phone station. For example, a PBX may be programmable to insert automatically the three-digit VOIP identifier strings into calls to be directed to analog VOIPs.

The MVP2410 offers complete flexibility for inter-operation with PBX units so that a coherent dialing scheme can be established to connect a company's multiple sites together in a way that is convenient and intuitive for phone users. When working together with modern PBX units, the presence of the MVP2410 can be completely transparent to phone users within the company.

Chapter 7: E1 Phonebook Configuration

(European Telephony Standards)

E1 Versus T1 Telephony Environments

Phonebooks for Series II analog MultiVOIP units (MVP130, MVP130FXS, MVP210, MVP410, and MVP810) can be operated in either a North American telephony standards environment (potentially operating with T1 digital MultiVOIPs) or in a European telephony standards environment (potentially operating with E1 digital MultiVOIPs). The configuration of the phonebook is the same in either case. However, because the telephony environment is different in each case and the examples used here must reflect those differences, we have separate chapters for phonebook configuration in North American (T1) environments (Chapter 6) and for that in European (E1) environments (Chapter 7). Consult the chapter that best fits the needs of your voip system. Phonebook configuration examples shown for Series II analog MultiVOIPs are fully applicable to ISDN-BRI MultiVOIPs except where the telephony interface is concerned.

E1-Standard Inbound and Outbound MultiVOIP Phonebooks

Important Definition:	The MultiVOIP's Outbound phonebook lists the phone stations it can call; its Inbound phonebook describes the dialing sequences that can be used to call that MultiVOIP and how those calls will be directed.

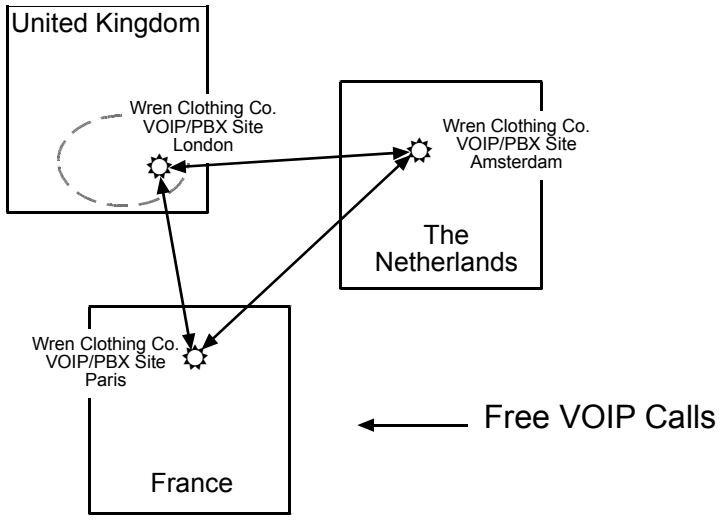
When a VOIP serves a PBX system, the operation of the VOIP should be transparent to the telephone end user and savings in long-distance calling charges should be enjoyed. Use of the VOIP should not require the dialing of extra digits to reach users elsewhere on the VOIP network. On the contrary, VOIP service more commonly reduces dialed digits by allowing users (served by PBXs in facilities in distant cities) to dial their co-workers with 3-, 4-, or 5-digit extensions -- as if they were in the same facility. More importantly, the VOIP system should be configured to maximize savings in long-distance calling charges. To achieve both of these objectives, ease of use and maximized savings, the VOIP phonebooks must be set correctly.

NOTE: VOIPs are commonly used for another reason, as well: VOIPs

allow an organization to integrate phone and data traffic onto a single network. Typically these are private networks.

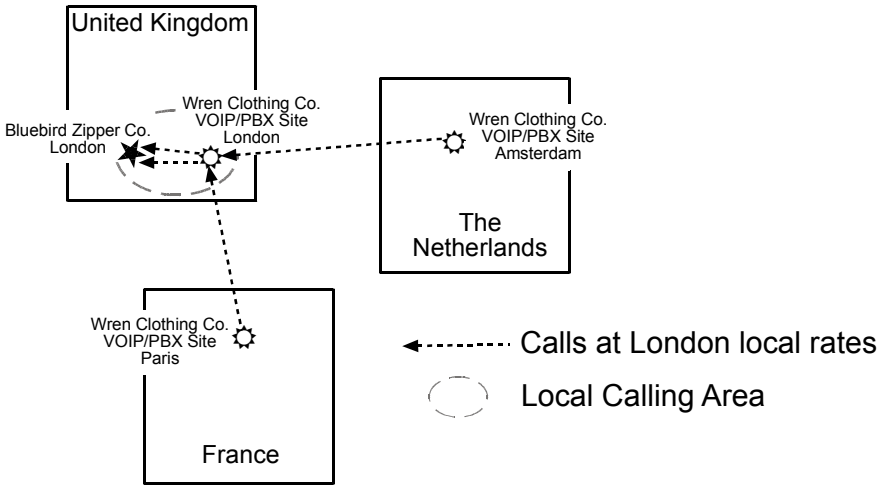
Free Calls: One VOIP Site to Another

The most direct use of the VOIP system is making calls between the offices where the VOIPs are located. Consider, for example, the Wren Clothing Company. This company has VOIP-equipped offices in London, Paris, and Amsterdam, each served by its own PBX. VOIP calls between the three offices completely avoid international long-distance charges. These calls are free. The phonebooks can be set up to allow all Wren Clothing employees to contact each other using 3-, 4-, or 5-digit numbers, as though they were all in the same building.

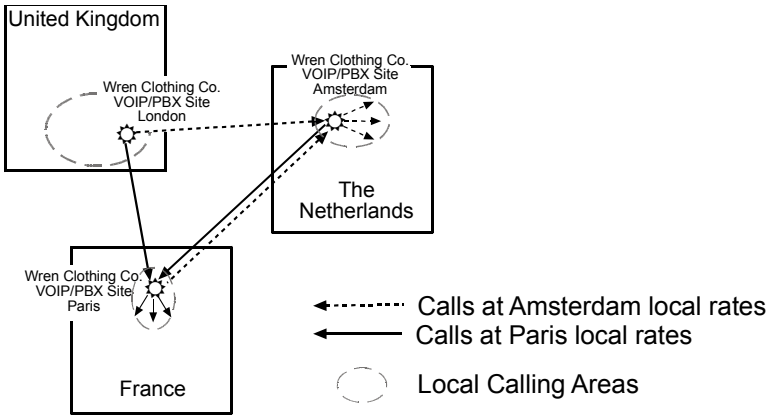


Local Rate Calls: Within Local Calling Area of Remote VOIP

In the second use of the VOIP system, the local calling area of each VOIP location becomes accessible to all of the VOIP system's users. As a result, international calls can be made at local calling rates. For example, suppose that Wren Clothing buys its zippers from The Bluebird Zipper Company in the western part of metropolitan London. In that case, Wren Clothing personnel in both Paris and Amsterdam could call the Bluebird Zipper Company without paying international long-distance rates. Only London local phone rates would be charged. This applies to calls completed anywhere in London's local calling area (which includes both Inner London and Outer London). Generally, local calling rates apply only within a single area code, and, for all calls outside that area code, national rates apply. There are, however, some European cases where local calling rates extend beyond a single area code. Local rates between Inner and Outer London are one example of this. (It is also possible, in some locations, that calls within an area code may be national calls. But this is rare.)

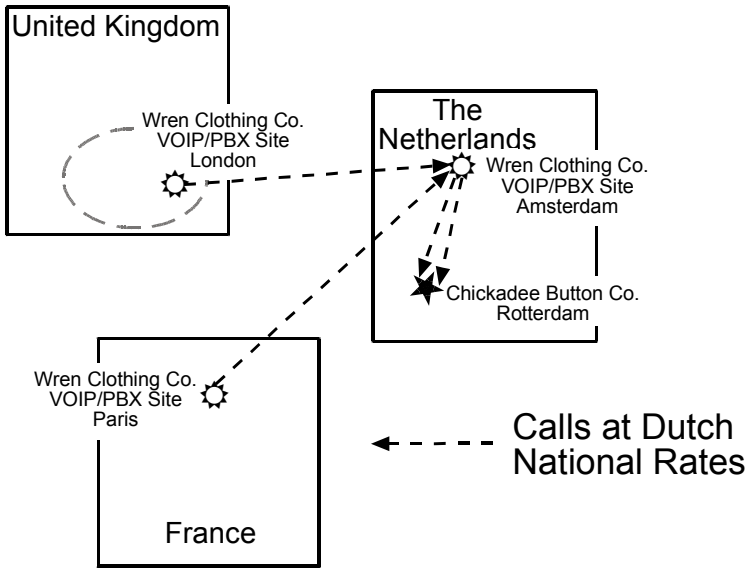


Similarly, the VOIP system allows Wren Clothing employees in London and Amsterdam to call anywhere in Paris at local rates; it allows Wren Clothing employees in Paris and London to call anywhere in Amsterdam at local rates.

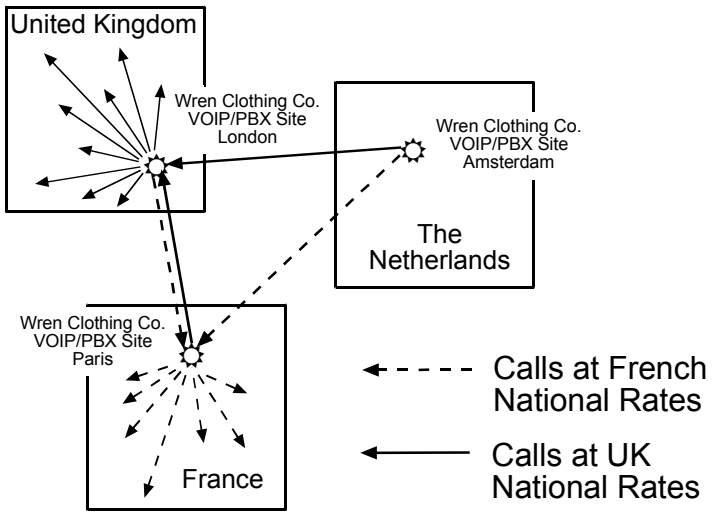


National Rate Calls: Within Nation of Remote VOIP Site

In the third use of the VOIP system, the national calling area of each VOIP location becomes accessible to all of the VOIP system's users. As a result, international calls can be made at national calling rates. Again, significant savings are possible. For example, suppose that the Wren Clothing Company buys its buttons from the Chickadee Button Company in the Dutch city of Rotterdam. In that case, Wren Clothing personnel in both London and Paris could call the Chickadee Button Company without paying international long-distance rates; only Dutch national calling rates would be charged. This applies to calls completed anywhere in The Netherlands.



Similarly, the VOIP system allows Wren Clothing employees in London and Amsterdam to call anywhere in France at French national rates; it allows Wren Clothing employees in Paris and Amsterdam to call anywhere in the United Kingdom at its national rates.



Inbound versus Outbound Phonebooks

To make the VOIP system transparent to phone users and to allow all possible free and reduced-rate calls, the VOIP administrator must configure the "Outbound" and "Inbound" phone-books of each VoIP in the system.

The "Outbound" phonebook for a particular VOIP unit describes the dialing sequences required for a call to originate locally (typically in a PBX in a particular facility) and reach any of its possible destinations at remote VOIP sites, including calls terminating at points beyond the remote VOIP site.

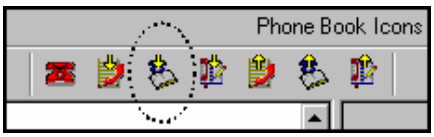
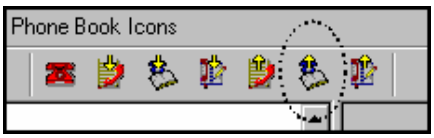
The "Inbound" phonebook for a particular VOIP unit describes the dialing sequences required for a call to originate remotely from any other VOIP sites in the system, and to terminate on that particular VOIP.

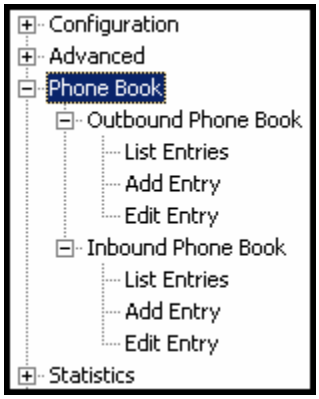
Briefly stated, *the MultiVOIP's Outbound phonebook lists the phone stations it can call; its Inbound phonebook lists the dialing sequences that can be used to call that MultiVOIP*. (Of course, the phone numbers are not literally "listed" individually.) The phone stations that can originate or complete calls over the VOIP system are described by numerical rules called "destination patterns." These destination patterns generally consist of country codes, area codes or city codes, and local phone exchange numbers.

In order for any VOIP phone call to be made, there must be both an Inbound Phonebook entry and an Outbound Phonebook entry that describe the end-to-end connection. The phone station originating the call must be connected to the VOIP system. The Outbound Phonebook for that VOIP unit must have a destination pattern entry that includes the 'called' phone (that is, the phone completing the call). The Inbound Phonebook of the VOIP where the call is completed must have a destination pattern entry that includes the digit sequence dialed by the originating phone station.

The PhoneBook Configuration procedure below is brief, but it is followed by an example case. For many people, the example case may be easier to grasp than the procedure steps. Configuration is not difficult, but all phone number sequences, destination patterns, and other information must be entered exactly; otherwise connections will not be made.

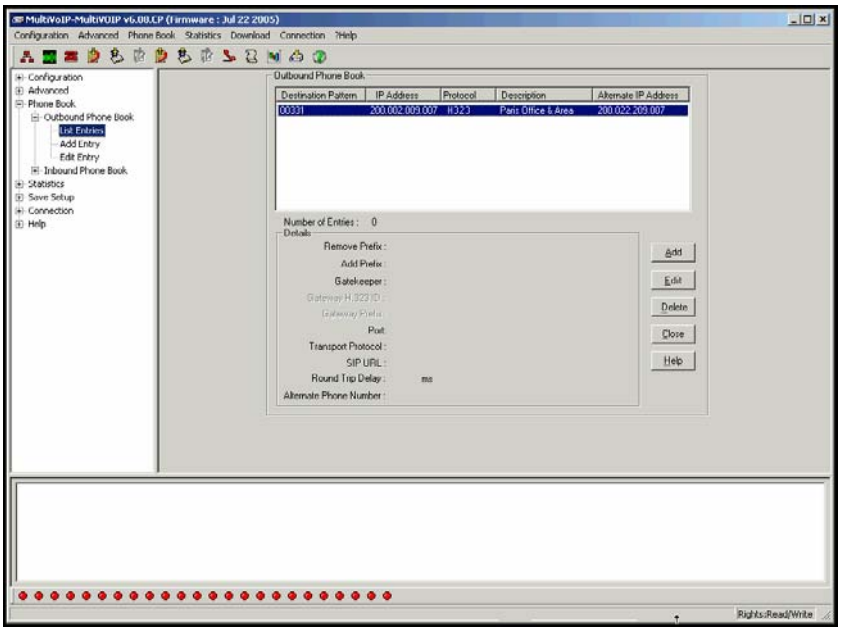
Phonebook configuration screens can be accessed using icons or the sidebar menu.

Phonebook Icons	Description
	Phonebook Configuration
	Inbound Phonebook Entries List
	Add Inbound Phonebook Entry
	Edit selected Inbound Phonebook Entry
	Outbound Phonebook Entries List
	Add Outbound Phonebook Entry
	Edit selected Outbound Phonebook Entry

Phonebook Pulldown Menu	
Inbound Phonebook Shortcut	**Outbound Phonebook Shortcut**
Alt + I	Alt + O
Phonebook Sidebar Menu	
	

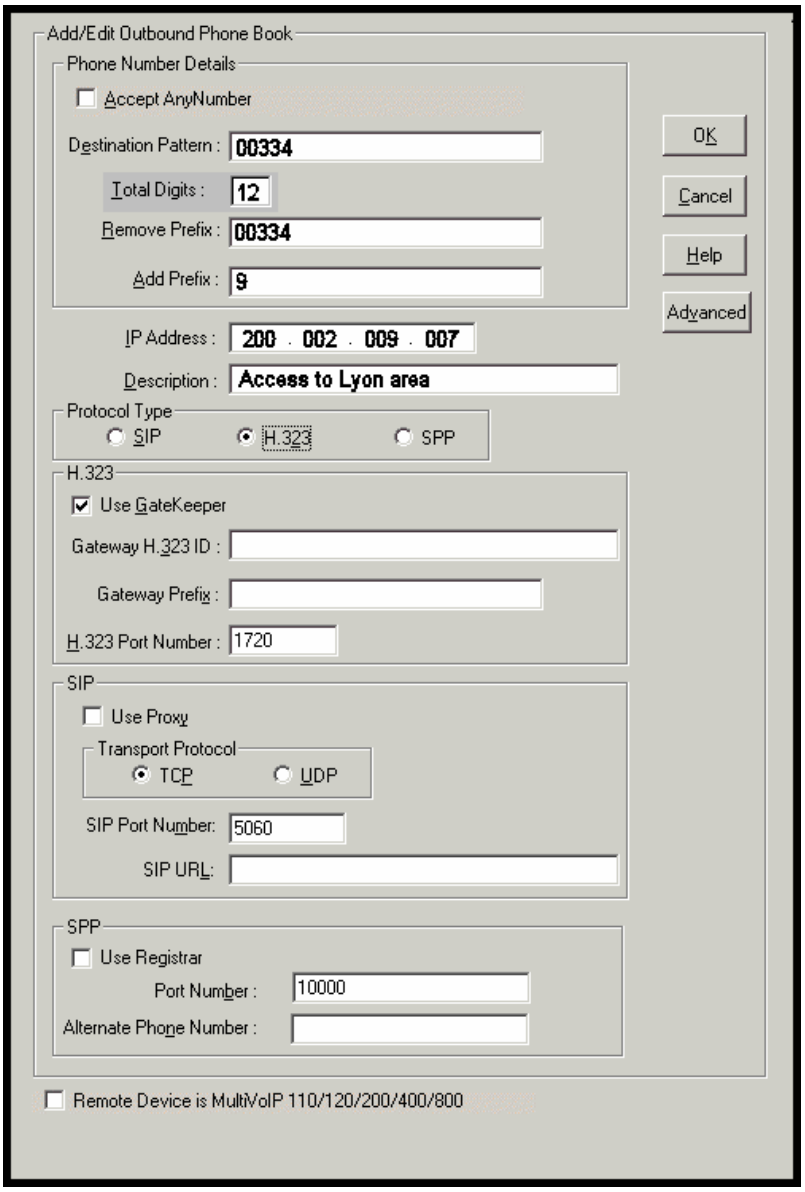
Phonebook Configuration Procedure

1. Select **Outbound Phone Book/List Entries**.



Click **Add**.

2. The **Add/Edit Outbound PhoneBook** screen appears.



Enter Outbound PhoneBook data for your MultiVOIP unit. Note that the Advanced button gives access to the Alternate IP Routing feature, if needed. Alternate IP Routing can be implemented in a secondary screen (as described after the primary screen field definitions below).

The fields of the **Add/Edit Outbound Phone Book** screen are described in the table below.

Add/Edit Outbound Phone Book: Field Definitions		
Field Name	**Values**	**Description**
Accept Any Number	Y/N	When checked, "Any Number" appears as the value in the Destination Pattern field. The **Any Number** feature works differently depending on whether or not an external routing device is used (Gatekeeper for H323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). **When no external routing device is used**. If **Any Number** is selected, calls to phone numbers not matching a listed Destination Pattern will be directed to the **IP Address** in the Add/Edit Outbound Phone Book screen. "Any Number" can be used in addition to one or more Destination Patterns. **When external routing device is used**. If Any Number is selected, calls to phone numbers not matching a listed Destination Pattern will be directed to the external routing device used (Gatekeeper for H323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). The IP Address of the external routing device must be set in the Phone Book Configuration screen.

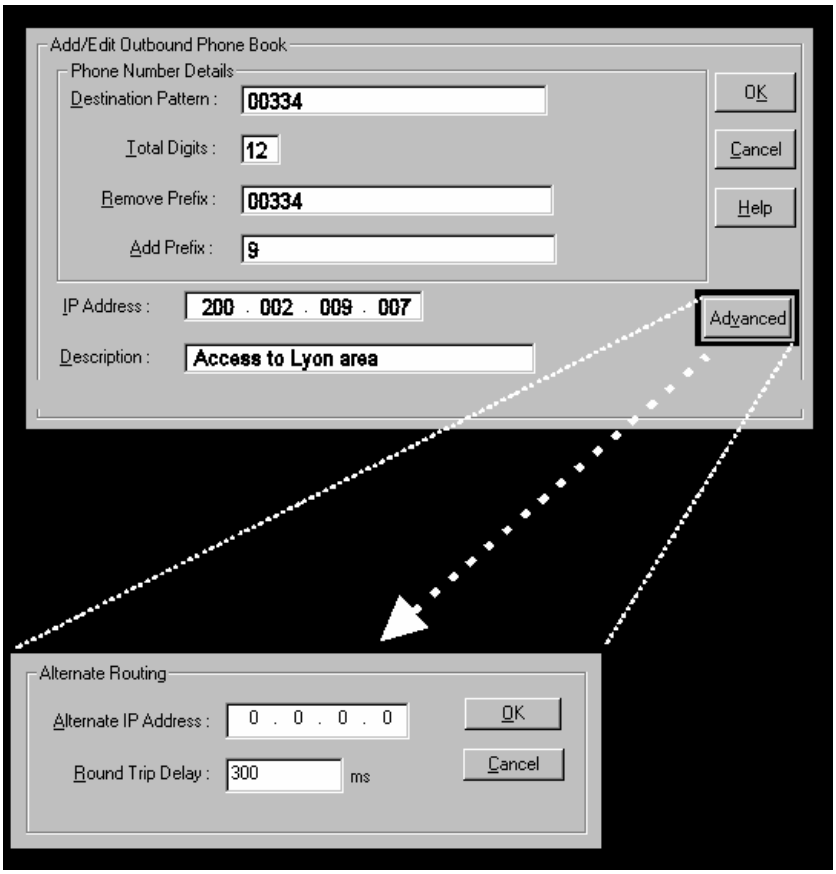
Add/Edit Outbound Phone Book: Field Definitions		
Field Name	**Values**	**Description**
Destination Pattern	prefixes, area codes, exchanges, line numbers, extensions	Defines the beginning of dialing sequences for calls that will be connected to another VOIP in the system. Numbers beginning with these sequences are diverted from the PTSN and carried on Internet or other IP network.
Total Digits	as needed	number of digits the phone user must dial to reach specified destination
Remove Prefix	dialed digits	portion of dialed number to be removed before completing call to destination
Add Prefix	dialed digits	digits to be added before completing call to destination
IP Address	n.n.n.n for = 0-255	the IP address to which the call will be directed if it begins with the destination pattern given
Description	alpha-numeric	Describes the facility or geographical location at which the call will be completed.
Protocol Type	SIP, H.323, or SPP	Indicates protocol to be used in outbound transmission.

Add/Edit Outbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
H.323 fields		
Use Gatekeepr	Y/N	Indicates whether or not gatekeeper is used.
Gateway H.323 ID	alpha-numeric	The H.323 ID assigned to the destination MultiVOIP. Only valid if "Use Gatekeeper" is enabled for this entry.
Gateway Prefix	numeric	This number becomes registered with the GateKeeper. Call requests sent to the gatekeeper and preceded by this prefix will be routed to the VOIP gateway.
H.323 Port Number	1720	This parameter pertains to Q.931, which is the H.323 call signaling protocol for setup and termination of calls (aka ITU-T Recommendation I.451). H.323 employs only one "well-known" port (1720) for Q.931 signaling. If Q.931 message-oriented signaling protocol is used, the port number 1720 must be chosen.

Add/Edit Outbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
SIP Fields		
Use Proxy	Y/N	Select if proxy server is used.
Transport Protocol	TCP or UDP	Voip administrator must choose between UDP and TCP transmission protocols. UDP is a high-speed, low-overhead connectionless protocol where data is transmitted without acknowledgment, guaranteed delivery, or guaranteed packet sequence integrity. TCP is slower connection-oriented protocol with greater overhead, but having acknowledgment and guarantees delivery and packet sequence integrity.
SIP Port Number	5060 or other *See RFC3087 ("Control of Service Context using SIP Request-URI," by the Network Working Group).	The SIP Port Number is a UDP logical port number. The voip will "listen" for SIP messages at this logical port. If SIP is used, 5060 is the default, standard, or "well known" port number to be used. If 5060 is not used, then the port number used is that specified in the SIP Request URI (Universal Resource Identifier).
SIP URL	*sip.userphone @ hostserver*, where "userphone" is the telephone number and "hostserver" is the domain name or an address on the network	Looking similar to an email address, a **SIP URL** identifies a user's address. In SIP communications, each caller or callee is identified by a SIP url: sip:user_name@host_name. The format of a sip url is very similar to an email address, except that the "sip:" prefix is used.

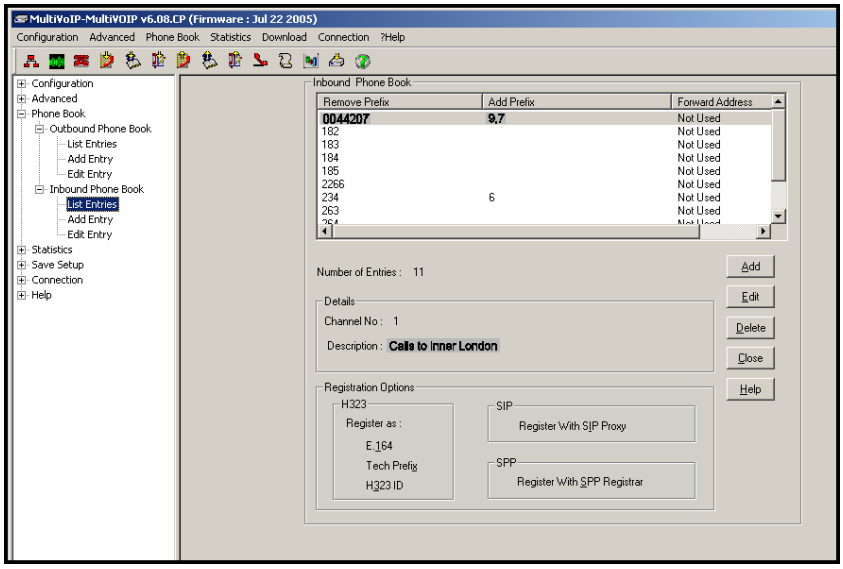
Add/Edit Outbound Phone Book: Field Def'ns (cont'd)		
Field Name	**Values**	**Description**
SPP Fields		
Use Registrar	**Values:** Y/N **Description:** Select this checkbox to use registrar when voip system is operating in the "Registrar/Client" SPP mode. In this mode, one voip (the registrar, as set in Phonebook Configuration screen) has a static IP address and all other voips (clients) point to the registar's IP address as functionally their own. However, if your voip system overall is operating in "Registrar/Client" mode but you want to make an exception and use Direct mode for the destination pattern of this particular Add/Edit Phonebook entry, leave this checkbox unselected. Leave this checkbox unselected if your overall voip system is operating in the "Direct" SPP mode. In this mode, all voips in system are peers and each has its own static IP address.	
Port Number	**Values:** numeric **Description:** When operating in "Registrar/Client" mode, this is the port by which the gateway receives all SPP data and control messages from the registrar gateway. (This ability to receive all data and messages via one port allows the voip to operate behind a firewall with only one port open.) When operating in "Direct" mode, this is the Port by which peer voips receive data and messages.	
Alternate Phone Number	numeric	Phone number associated with alternate IP routing.
Remote Device is …	Y/N	Check when system includes 1st-generation MultiVOIPs to allow inter-operation. These include MVP-110/120/200/400/800 MultiVOIP units.
Advanced button	**Values:** N/A **Description:** Gives access to secondary screen where an Alternate IP Route can be specified for backup or redundancy of signal paths. See discussion on next page. For SIP & H.323 operation only.	

Clicking on the **Advanced** button brings up the **Alternate Routing** secondary screen. This feature provides an alternate path for calls if the primary IP network cannot carry the traffic. Often in cases of failure, call traffic is temporarily diverted into the PSTN. However, this feature could also be used to divert traffic to a redundant (backup) unit in case one voip unit fails. The user must specify the IP address of the alternate route for each destination pattern entry in the Outbound Phonebook.

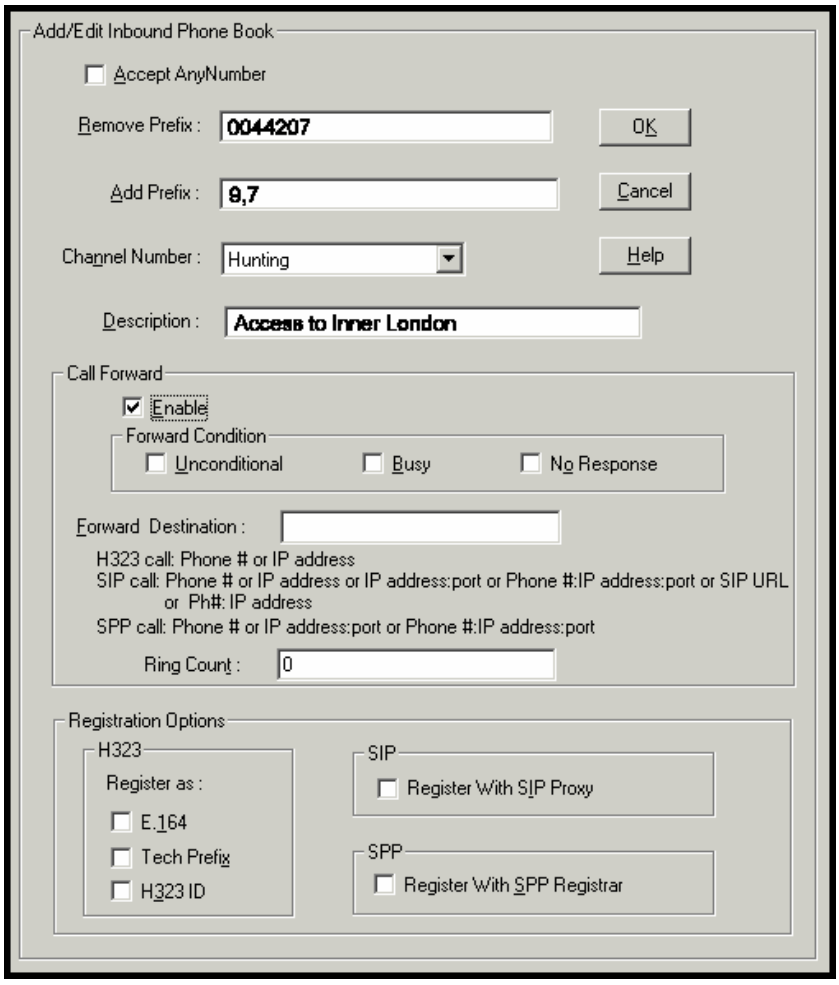


Alternate Routing Field Definitions		
Field Name	**Values**	**Description**
Alternate IP Address	n.n.n.n where n= 0-255	Alternate destination for outbound data traffic in case of excessive delay in data transmission.
Round Trip Delay	milliseconds	The Round Trip Delay is the criterion for judging when a data pathway is considered blocked. When the delay exceeds the threshold specified here, the data stream will be diverted to the alternate destination specified as the Alternate IP Address.

3. Select **Inbound PhoneBook/List Entries**.



4. The **Add/Edit Inbound PhoneBook** screen appears.



Enter Inbound PhoneBook data for your MultiVOIP unit. The fields of the Add/Edit Inbound PhoneBook screen are described in the table below.

Add/Edit Inbound Phone Book: Field Definitions		
Field Name	**Values**	**Description**
Accept Any Number	Y/N	When checked, "Any Number" appears as the value in the Remove Prefix field. The **Any Number** feature of the Inbound Phone Book does not work when an external routing device is used (Gatekeeper for H323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). **When no external routing device is used**. If **Any Number** is selected, calls received from phone numbers not matching a listed Prefix (shown in the Remove Prefix column of the Inbound Phone Book) will be admitted into the voip on the channel listed in the **Channel Number** field. "Any Number" can be used in addition to one or more Prefixes.
Remove Prefix	dialed digits	portion of dialed number to be removed before completing call to destination (often a local PBX)
Add Prefix	dialed digits	digits to be added before completing call to destination (often a local PBX)

Add/Edit Inbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Channel Number	1-30, or "Hunting"	E1 channel number to which the call will be assigned as it enters the local telephony equipment (often a local PBX). "Hunting" directs the call to any available channel.
Description	--	Describes the facility or geographical location at which the call originated.
Call Forward Parameters		
Enable	Y/N	Click the check-box to enable the call-forwarding feature.
Forward Condition	Uncondit.; Busy No Resp.	**Unconditional**. When selected, all calls received will be forwarded. **Busy**. When selected, calls will be forwarded when station is busy. **No Response**. When selected, calls will be forwarded if called party does not answer after a specified number of rings, as specified in **Ring Count** field. Forwarding can be conditioned on both "Busy" and "No Response."

Add/Edit Inbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Forward Destination IP address, phone number, port number, etc.	Phone number or IP address to which calls will be directed. For H.323 calls, the Forward Destination can be either a Phone Number of an IP Address. For SIP calls, the Forward Destination can be one of the following: (a) phone number, (b) IP address, (c) IP address: port number, (d) phone number:IP addr: port number, (e) SIP URL, or (f) phone #: IP address. For SPP calls, the Forward Destination can be one of the following: (a) phone number, (b) IP address: port, or (c) phone number: IP address: port.	
Ring Count	integer	When No Response is condition for forwarding calls, this determines how many unanswered rings are needed to trigger the forwarding.
Registration Option Parameters	In an H.323 voip system, gateways can register with the system using one of these identifiers: (a) an E.164 identifier, (b) a Tech Prefix identifier, or (c) an H.323 ID identifier. In a SIP voip system, gateways can register with the SIP Proxy. In an SPP voip system, gateways can register with the SPP Registrar voip unit.	

5. When your Outbound and Inbound PhoneBook entries are completed, click on **Save Setup** in the sidebar menu to save your configuration.

   You can change your configuration at any time as needed for your system.

Remember that the initial MultiVOIP setup must be done locally or via the built-in Remote Configuration/Command Modem using the MultiVOIP program. However, after the initial configuration is complete, all of the MultiVOIP units in the VOIP system can be configured, re-configured, and updated from one location using the MultiVOIP web GUI software program or the MultiVOIP program (in conjunction with the built-in modem).

E1 Phonebook Examples

To demonstrate how Outbound and Inbound PhoneBook entries work in an international VOIP system, we will re-visit our previous example in greater detail. It's an international company with offices in London, Paris, and Amsterdam. In each office, a MVP3010 has been connected to the PBX system.

3 Sites, All-E1 Example

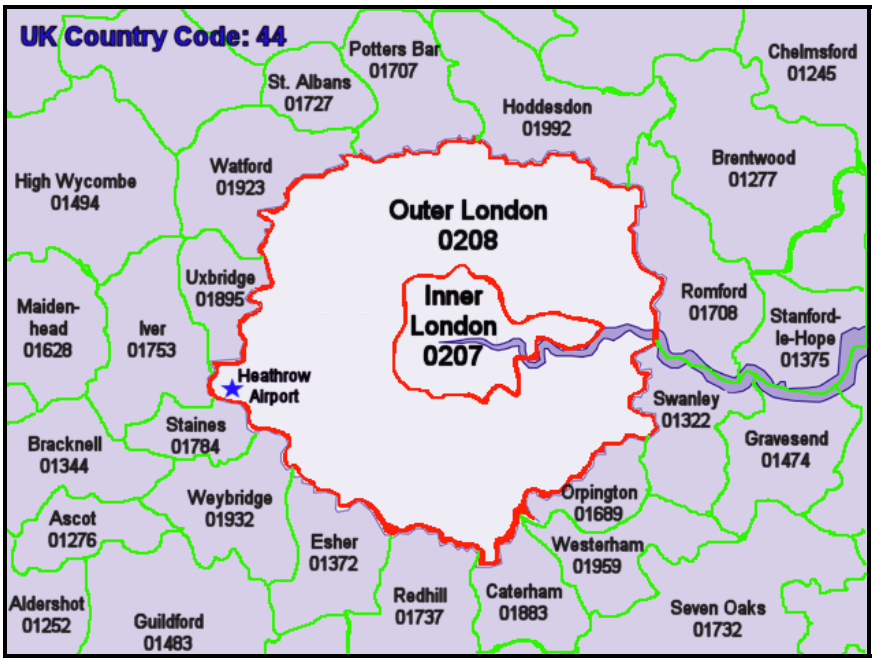
The VOIP system will have the following features:

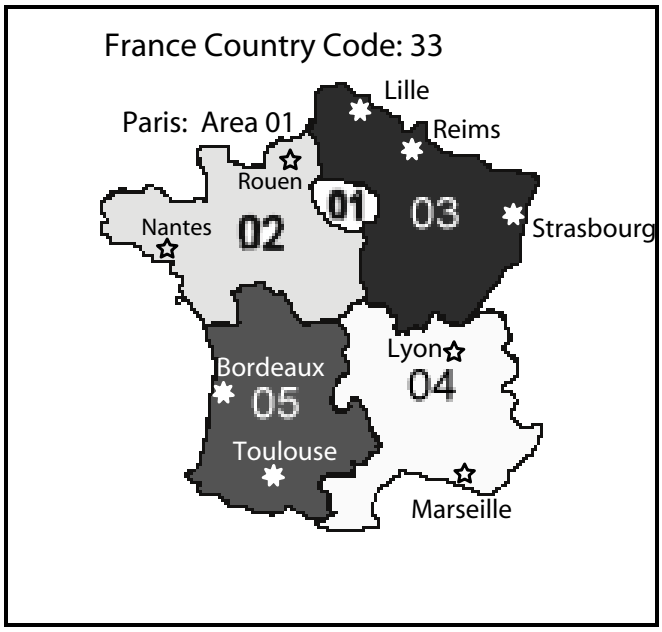
1. Employees in all cities will be able to call each other over the VOIP system using 4-digit extensions.

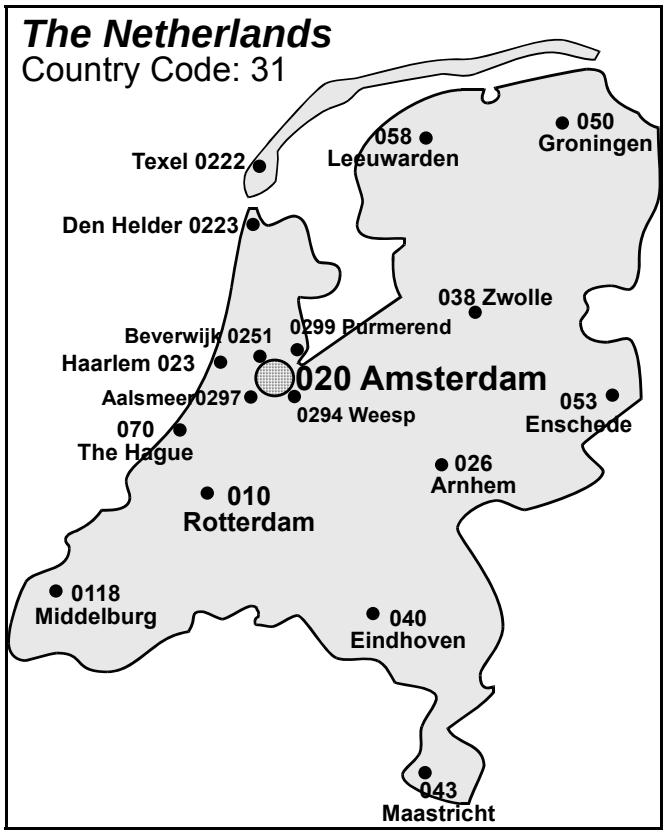
2. Calls to Outer London and Inner London, greater Amsterdam, and greater Paris will be accessible to all company offices as local calls.

3. Vendors in Guildford, Lyon, and Rotterdam can be contacted as national calls by all company offices.

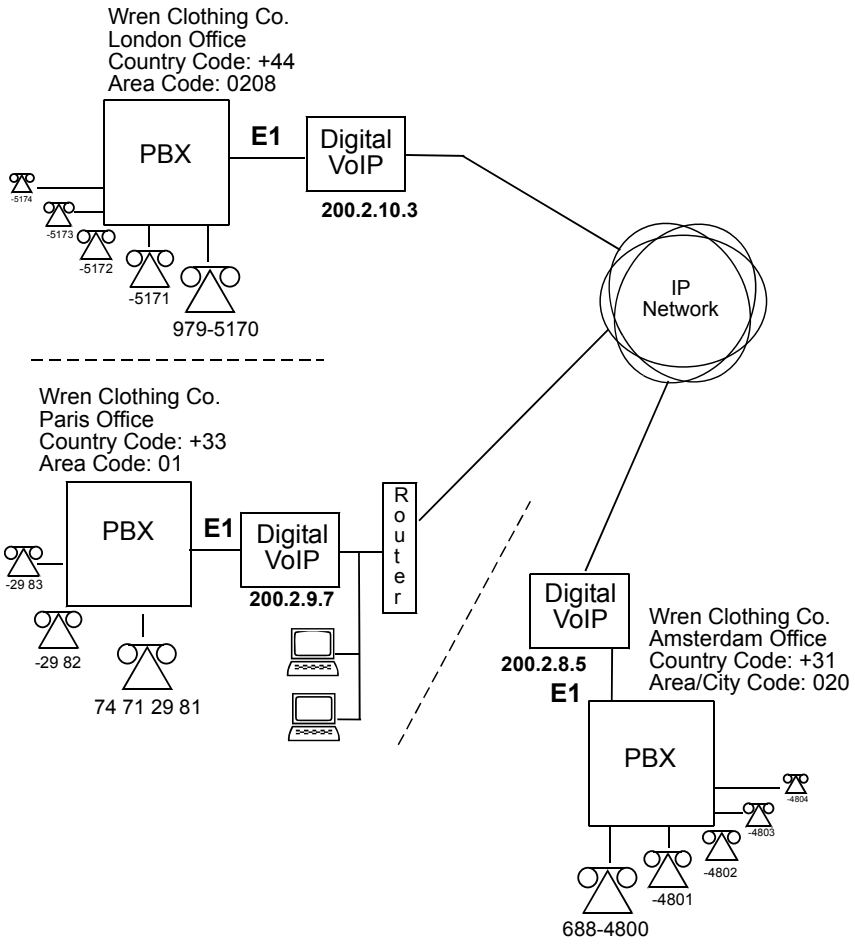
Note that the phonebook entries for Series II analog MultiVOIPs (MVP-210/410/810) used in Euro-type telephony settings will be the same in format as entries for the MVP3010.


UK Country Code: 44
Potters Bar 01707
St. Albans 01727
Hoddesdon 01992
Chelmsford 01245
Brentwood 01277
High Wycombe 01494
Watford 01923
Outer London 0208
Inner London 0207
Uxbridge 01895
Maiden-head 01628
Iver 01753
Heathrow Airport
Romford 01708
Stanford-le-Hope 01375
Swanley 01322
Gravesend 01474
Staines 01784
Bracknell 01344
Weybridge 01932
Orpington 01689
Ascot 01276
Esher 01372
Westerham 01959
Redhill 01737
Caterham 01883
Seven Oaks 01732
Aldershot 01252
Guildford 01483

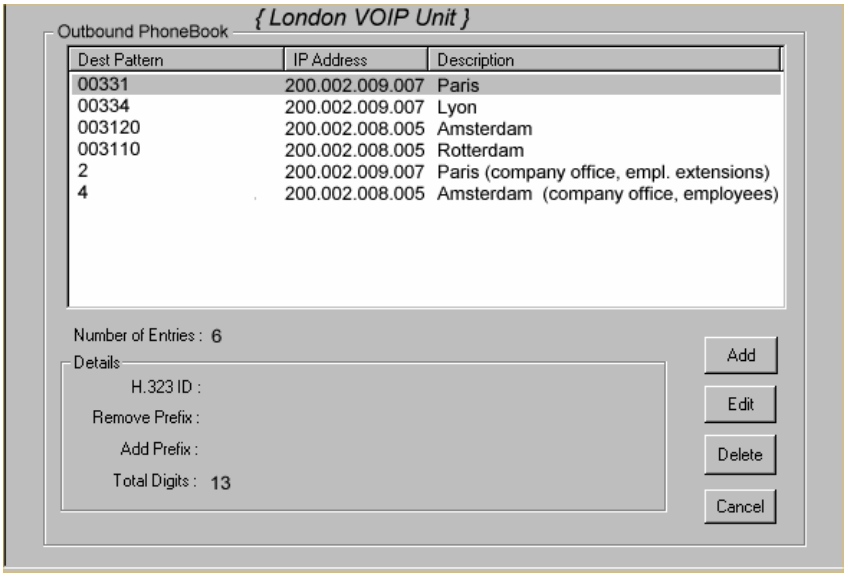

France Country Code: 33
Lille
Paris: Area 01
Reims
Rouen
01
03
Strasbourg
Nantes
02
Lyon
Bordeaux
04
05
Toulouse
Marseille


The Netherlands
Country Code: 31
050
Groningen
058
Leeuwarden
Texel 0222
Den Helder 0223
038 Zwolle
0299 Purmerend
Beverwijk 0251
Haarlem 023
020 Amsterdam
Aalsmeer 0297
0294 Weesp
053
Enschede
070
The Hague
026
Arnhem
010
Rotterdam
0118
Middelburg
040
Eindhoven
043
Maastricht

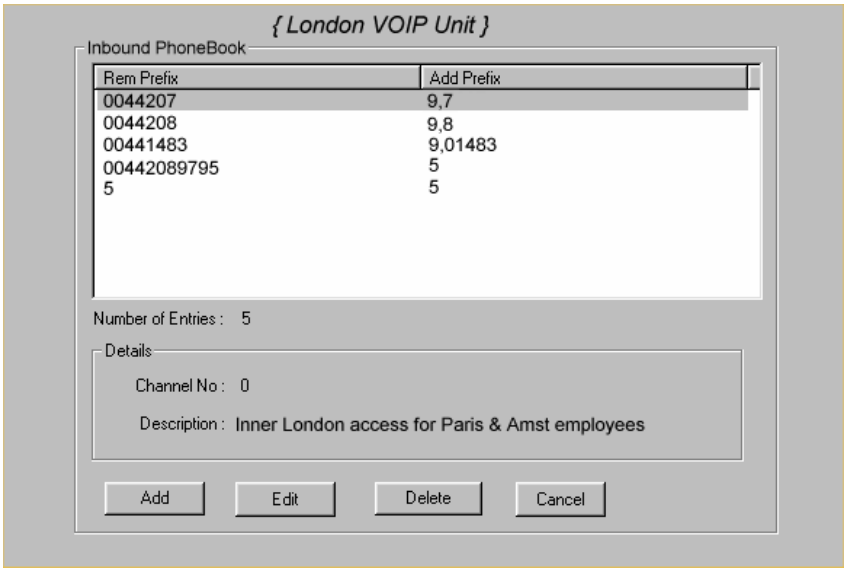
An outline of the equipment setup in these three offices is shown below.



The screen below shows Outbound PhoneBook entries for the VOIP located in the company's London facility

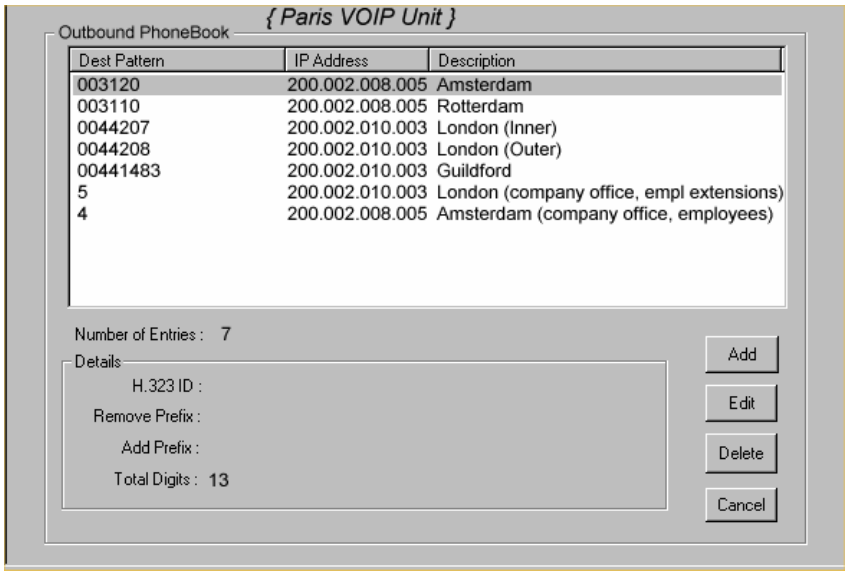


The Inbound PhoneBook for the London VOIP is shown below.

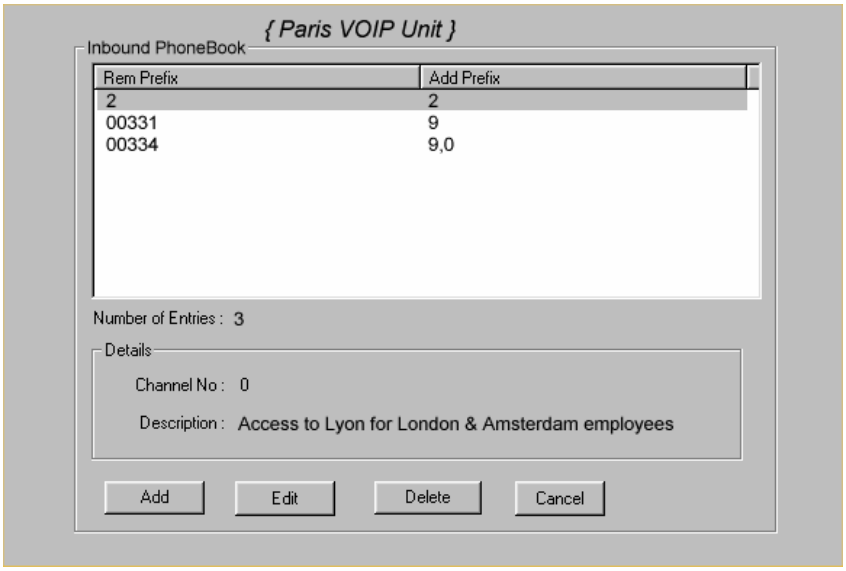


NOTE: Commas are allowed in the Inbound Phonebook, but **not** in the Outbound Phonebook. Commas denote a brief pause for a dial tone, allowing time for the PBX to get an outside line.

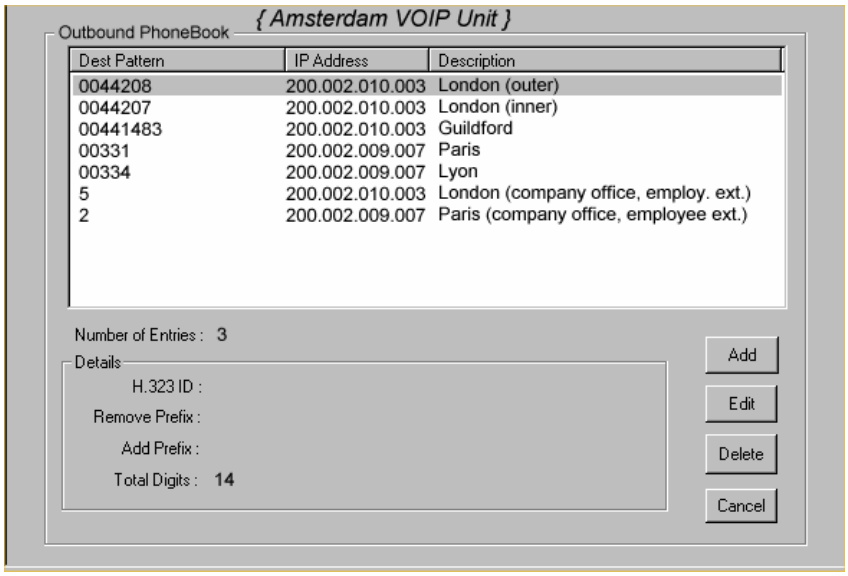
The screen below shows Outbound PhoneBook entries for the VOIP located in the company's Paris facility.



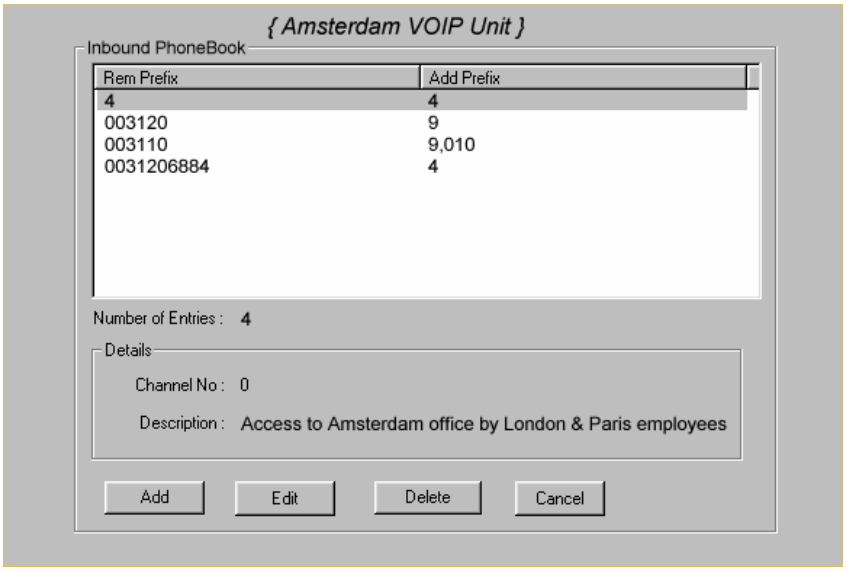
The Inbound PhoneBook for the Paris VOIP is shown below.



The screen below shows Outbound PhoneBook entries for the VOIP in the company's Amsterdam facility.

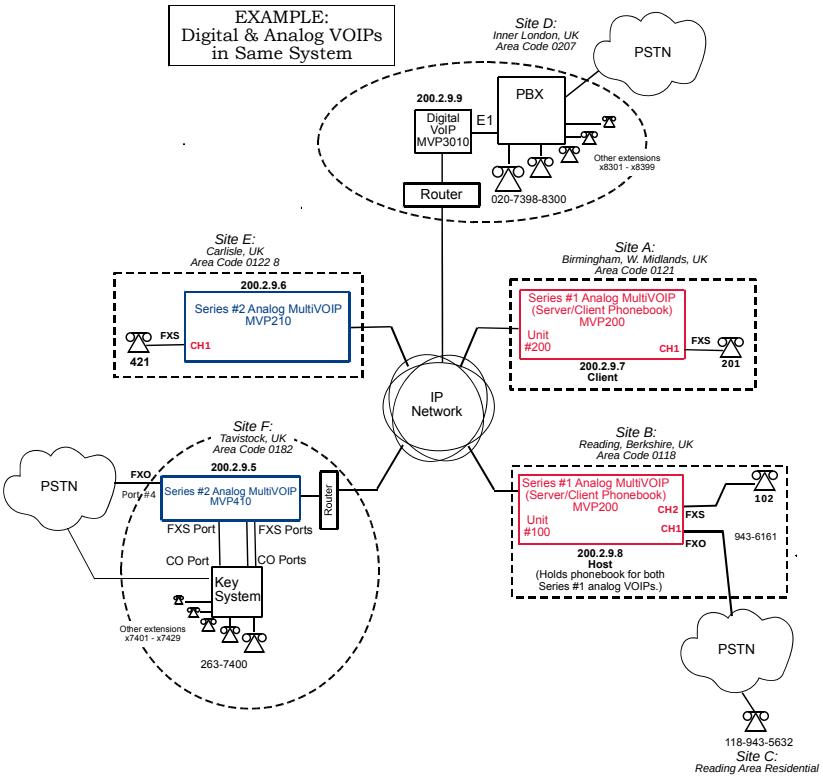


The Inbound PhoneBook for the Amsterdam VOIP is shown below.



Configuring Digital & Analog VOIPs in Same System

Analog MultiVOIP units, like the MVP-210/410/810 are compatible with digital MultiVOIP units like the MVP3010. In many cases, digital and analog VOIP units will appear in the same telephony/IP system. In addition to MVP-210/410/810 MultiVOIP units (Series II units), legacy analog VOIP units (Series I units made by MultiTech) may be included in the system, as well. When legacy VOIP units are included, the VOIP administrator must handle two styles of phonebooks in the same VOIP network. The diagram below shows a small-scale system of this kind: one digital VOIP (the MVP3010) operates with two Series II analog VOIPs (an MVP210 and an MVP410), and two Series I legacy VOIPs (two MVP200 units).



The Series I analog VOIP phone book resides in the "Host" VOIP unit at Site B. It applies to both of the Series I analog VOIP units.

Each of the Series II analog MultiVOIPs (the MVP210 and the MVP410) requires its own inbound and outbound phonebooks. The MVP3010 digital MultiVOIP requires its own inbound and outbound phonebooks, as well.

These **seven** phone books are shown below.

Phone Book for Analog VOIP Host Unit (Site B)			
VOIP Dir # -OR- Destination Pattern	IP Address	Channel	Comments
102	200.2.9.8	2	Site B, FXS channel. (Reading, UK)
101	200.2.9.8	1	Site B, FXO channel. (Reading, UK)
201	200.2.9.7	1	Site A, FXS channel. (Birmingham)
421	200.2.9.6	0	Site E, FXS channel. (Carlisle, UK)
018226374 Note 3.	200.2.9.5	0	Gives remote voip users access to key phone system extensions at Tavistock office (Site F). The key system might be arranged either so that calls go through a human operator or through an auto-attendant (which prompts user to dial the desired extension).
0182	200.2.9.5	4	Gives remote voip users access to Tavistock PSTN via FXO port (#4) at Site F.
3xx	200.2.9.9	0 (Note 1.)	Allows remote voip users to call all PBX extensions at Site D (Inner London) using only three digits.

Phone Book for Analog VOIP Host Unit (Site B) (continued)			
VOIP Dir # -OR- Destination Pattern	IP Address	Channel	Comments
0207 xxx xxxx	200.2.9.9	0 (Note 2.)	Gives remote voip users access to phone numbers in 0207 area code (Inner London) in which Site D is located.
0208 xxx xxxx	200.2.9.9	0 (Note 2.)	Gives remote voip users access to phone numbers in 0208 area code (Outer London) for which calls are local from Site D (Inner London).

Note 1. The "x" is a wildcard character.

Note 2. By specifying "Channel 0," we instruct the MVP3010 to choose any available data channel to carry the call.

Note 3. Note that Site F key system has only 30 extensions (x7400-7429). This destination pattern (018226374) actually directs calls to 402-263-74**30** through 402-263-74**99** into the key system, as well.
This means that such calls, which belong on the PSTN, cannot be completed. In some cases, this might be inconsequential because an entire exchange (fully used or not) might have been reserved for the company or it might be unnecessary to reach those numbers. However, to specify only the 30 lines actually used by the key system, the destination pattern 018226374 would have to be replaced by three other destination patterns, namely 0182263740, 0182263741, and 0182263742. In this way, calls to 0182-263-74**30** through 0182-263-74**99** would be properly directed to the PSTN. In the Site D outbound phonebook, the 30 lines are defined exactly, that is, without making any adjacent phone numbers unreachable through the voip system.

The Outbound PhoneBook of the MVP3010 is shown below.

Outbound Phone Book for MVP3010 Digital VOIP (Site D)				
Destin. Pattern	**Remove Prefix**	**Add Prefix**	**IP Address**	**Comment**
201			200.2.9.7	To originate calls to Site A (Birmingham).
901189	901189	101# Note 3.	200.2.9.8	To originate calls to any PSTN phone in Reading area using the FXO channel (channel #1) of the Site B VOIP (Reading, UK).
421	--	--	200.2.9.6	Calls to Site E (Carlisle).
90182				Calls to Tavistock local PSTN (Site F) could be arranged by operator or possibly by auto-attendant.
90182 263 740	9	--	200.2.9.5	Calls to extensions of key phone system at Tavistock office.
90182 263 741	9	--	200.2.9.5	
90182 263 742	9	--	200.2.9.5	
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP (Reading).
Note 3. The pound sign (“#”) is a delimiter separating the VOIP number from the standard telephony phone number.				

The Inbound PhoneBook of the MVP3010 is shown below.

Inbound Phone Book for MVP3010 Digital VOIP (Site D)			
Remove Prefix	**Add Prefix**	**Channel Number**	**Comments**
0207	9,7 Note 4. Note 5.	0	Allows phone users at remote voip sites to call local numbers (those within the Site D area code, 0207, Inner London) over the VOIP network.
0208	9,8 Note 4. Note 5.	0	Allows phone users at remote voip sites to call local numbers (those in Outer London) over the VOIP network.
0207 39883	3	0	Allows phone users at remote voip sites to call extensions of the Site D PBX using three digits, beginning with "3" .
Note 4. "9" gives PBX station users access to outside line. Note 5. The comma represents a one-second pause, the time required for the user to receive a dial tone on the outside line (PSTN). Commas can be used in the Inbound Phonebook, but **not** in the Outbound Phonebook.			

Outbound Phone Book for MVP410 Analog VOIP (Site F)				
Destin. Pattern	Remove Prefix	**Add Prefix**	**IP Address**	**Comment**
201			200.2.9.7	To originate calls to Site A (Birmingham).
01189	0118	101# Note 3.	200.2.9.8	To originate calls to any PSTN phone in Reading area using the FXO channel (channel #1) of the Site B VOIP.
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP (Reading).
421			200.2.9.6	Calls to Site E (Carlisle).
0207			200.2.9.9	Calls to Inner London area PSTN via Site D PBX.
0208			200.2.9.9	Calls to Inner London area PSTN via Site D PBX.
3	--	0207 398 8	200.2.9.9	Calls to Inner London PBX extensions with three digits.
Note 3. The pound sign ("#") is a delimiter separating the VOIP number from the standard telephony phone number.				

Inbound Phonebook for MVP410 Analog VOIP (Site F)			
Remove Prefix	**Add Prefix**	**Channel Number**	**Comment**
01822	2	4	Calls to Tavistock local PSTN through FXO port (Port #4) at Site F.
0182 263 740	740.	0	Gives remote voip users, access to extensions of key phone system atTavistock office. Because call is completed at key system, abbreviated dialing (3-digits) is not workable. Human operator or auto-attendant is needed to complete these calls.
0182 263 741	741.	0	
0182 263 742	742	0	

Outbound Phone Book for MVP210 Analog VOIP (Site E)				
Destin. Pattern	Remove Prefix	**Add Prefix**	**IP Address**	**Comment**
201			200.2.9.7	To originate calls to Site A (Birmingham).
01189	0118	101# Note 3.	200.2.9.8	To originate calls to any PSTN phone in Reading area using the FXO channel (channel #1) of the Site B VOIP.
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP (Reading).
01822	01822	--	200.2.9.5	Calls to Tavistock area PSTN (via FXO channel of the Site F VOIP).
0182 26374			200.2.9.5	Calls to Tavistock key system operator or auto-attendant.
0207	0207		200.2.9.9	Calls to London area PSTN via Site D PBX.
8		0207 398	200.2.9.9	Calls to London PBX extensions with four digits.
Note 3. The pound sign ("#") is a delimiter separating the VOIP number from the standard telephony phone number.				

Inbound Phonebook for MVP210 Analog VOIP (Site E)			
Remove Prefix	**Add Prefix**	**Channel Number**	**Comment**
421		1	

Call Completion Summaries

Site A calling Site C, Method 1

1. Dial 101.
2. Hear dial tone from Site B.
3. Dial 9435632.
4. Await completion. Talk.

Site A calling Site C, Method 2

5. Dial 101#9435632
6. Await completion. Talk.

Note: Some analog VOIP gateways will allow completion by Method 2. Others will not.

Site C calling Site A

1. Dial 9436161.
2. Hear dial tone from Site B VOIP.
3. Dial 201.
4. Await completion. Talk.

Site D calling Site C

1. Dial 901189435632.
2. "9" gets outside line. On some PBXs, an "8" may be used to direct calls to the VOIP, while "9" directs calls to the PSTN. However, some PBX units can be programmed to identify the destination patterns of all calls to be directed to the VOIP.
3. PBX at Site D is programmed to divert all calls made to the 118 area code and exchange 943 into the VOIP network. (It would also be possible to divert *all* calls to all phones in area code 118 into the VOIP network, but it may not be desirable to do so.)
4. The MVP3010 removes the prefix "0118" and adds the prefix "101#" for compatibility with the analog MultiVOIP's phonebook scheme. The "#" is a delimiter separating the analog VOIP's phone number from the digits that the analog VOIP must dial onto its local PSTN to complete the call. The digits "101#9435632" are forwarded to the Site B analog VOIP.
5. The call passes through the IP network (in this case, the Internet).
6. The call arrives at the Site B VOIP. This analog VOIP receives this dialing string from the MVP3010: 101#9435632. The analog VOIP, seeing the "101" prefix, uses its own channel #1 (an FXO port) to connect the call to the PSTN. Then the analog VOIP dials its local phone number 9435632 to complete the call.

NOTE: In the case of Reading, Berkshire,, England, both "1189" and "1183" are considered local area codes. This is, in a sense however, a matter of terminology. It simply means that numbers of the form 9xx-xxxx and 3xx-xxxx are both local calls for users at other sites in the VOIP network.

Site D calling Site F

A voip call from Inner London PBX to extension 7424 on the key telephone system in Tavistock, UK.

A. The required entry in the London Outbound Phonebook to facilitate origination of the call, would be 90182263742. The call would be directed to the Tavistock voip's IP address, 200.2.9.5. (Generally on such a call, the caller would have to dial an initial "9". But typically the PBX would not pass the initial "9" dialed to the voip. If the PBX *did* pass along that "9" however, its removal would have to be specified in the local Outbound Phonebook.)

B. The corresponding entry in the Tavistock Inbound Phonebook to facilitate completion of the call would be

0182263742	for calls within the office at Tavistock
01822	for calls to the Tavistock local calling area (PSTN).

Call Event Sequence

1. Caller in Inner London dials 901822637424.
2. Inner London voip removes "9" .
3. Inner London voip passes remaining string, 01822637424on to the Tavistock voip
   at IP address 200.2.9.5.
4. The dialed string matches an inbound phonebook entry at the Tavistock voip, namely 0182263742.
5. The Tavistock voip rings one of the three FXS ports connected to the Tavistock
   key phone system.
6. The call will be routed to extension 7424 either by a human receptionist/
   operator or to an auto-attendant (which allows the caller to specify the
   extension to which they wish to be connected).

Site F calling Site D

A voip call from a Tavistock key extension to extension 3117 on the PBX in Inner London.

A. The required entry in the Tavistock Outbound Phonebook to facilitate origination of the call, would be "3". The string 02073988 is added, preceding the "3". The call would be directed to the Inner London voip's IP address, 200.2.9.9.

B. The corresponding entry in the Inner-London Inbound Phonebook to facilitate completion of the call would be 020739883.

1. The caller in Tavistock picks up the phone receiver, presses a button on the key phone set. This button has been assigned to a particular voip channel.
2. The caller in Tavistock hears dial tone from the Tavistock voip.
3. The caller in Tavistock dials 02073983117.
4. The Tavistock voip sends the entire dialed string to the Inner-London voip
   at IP address 200.2.9.9.
5. The Inner-London voip matches the called digits 02073983117to its Inbound Phonebook entry "020739883, " which it removes. Then it adds back the "3" as a prefix.
6. The Inner-London PBX dials extension 3117 in the office in Inner London.

Variations in PBX Characteristics

The exact dialing strings needed in the Outbound and Inbound Phonebooks of the MVP3010 will depend on the capabilities of the PBX. Some PBXs require trunk access codes (like an "8" or "9" to access an outside line or to access the VOIP network). Other PBXs can automatically distinguish between intra-PBX calls, PSTN calls, and VOIP calls.

Some PBX units can also insert digits automatically when they receive certain dialing strings from a phone station. For example, a PBX may be programmable to insert automatically the three-digit VOIP identifier strings into calls to be directed to analog VOIPs.

The MVP3010 offers complete flexibility for inter-operation with PBX units so that a coherent dialing scheme can be established to connect a company's multiple sites together in a way that is convenient and intuitive for phone users. When working together with modern PBX units, the presence of the MVP3010 can be completely transparent to phone users within the company.

International Telephony Numbering Plan Resources

Due to the expansion of telephone number capacity to accommodate pagers, fax machines, wireless telephony, and other new phone technologies, numbering plans have been changing worldwide. Many new area codes have been established; new service categories have been established (for example, to accommodate GSM, personal numbering, corporate numbering, etc.). Below we list several web sites that present up-to-date information on the telephony numbering plans used around the world. While we find these to be generally good resources, we would note that URLs may change or become nonfunctional, and we cannot guarantee the quality of information on these sites.

URL	Description
http://phonebooth.interocitor.net/wtng	The World Telephone Numbering Guide presents excellent international numbering info that is both broad and detailed. This includes info on re-numbering plans carried out worldwide in recent years to accommodate new technologies.
http://www.oftel.gov.uk/numbers/number.htm	UK numbering plan from the Office of Telecommunications, the UK telephony authority.
http://www.itu.int/home/index.html	The International Telecommunications Union is an excellent source and authority on international telecom regulations and standards. National and international number plans are listed on this site.

URL	Description
http://kropla.com/phones.htm	Guide to international use of modems.
http://www.numberplan.org/	National and international numbering plans based on direct input from regulators worldwide. Includes lists of telecom carriers per country.
http://www.eto.dk/	European Telecommunications Office. Primarily concerned with mobile/wireless radiotelephony, GSM, etc.
http://www.eto.dk/ETNS.htm	European Telephony Numbering Space. Resources for pan-European telephony services, standards, etc. Part of ETO site.
http://www.regtp.de/en/reg_tele/start/fs_05.html	List of European telecom regulatory agencies by country (from German telecom authority).

Chapter 8: Operation and Maintenance

Operation and Maintenance

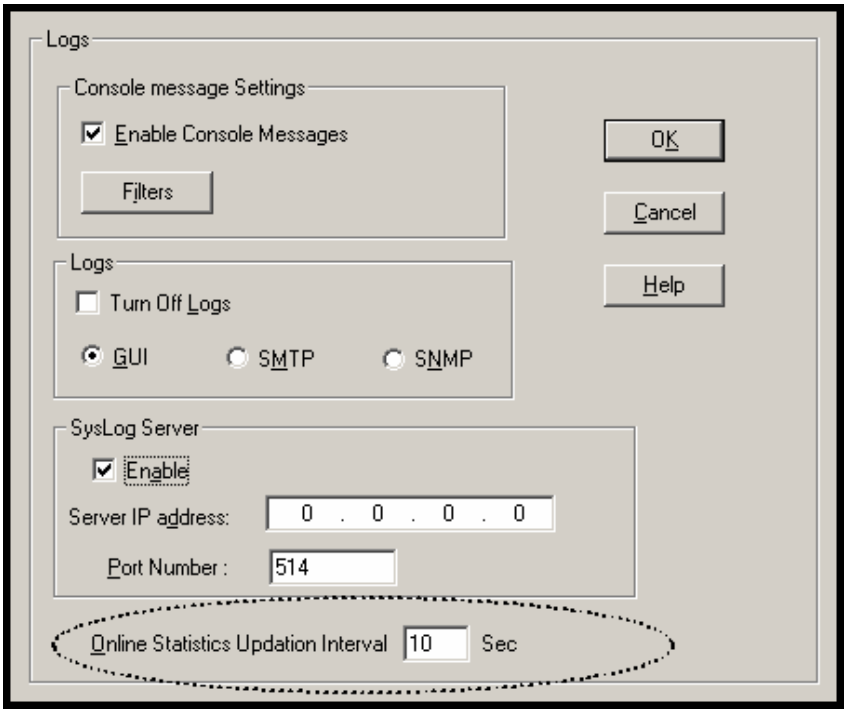
Although most Operation and Maintenance functions of the software are in the **Statistics** group of screens, an important summary appears in the **System Information** of the **Configuration** screen group.

System Information screen

This screen presents vital system information at a glance. Its primary use is in troubleshooting. This screen is accessible via the **Configuration** pulldown menu, the **Configuration** sidebar menu, or by the keyboard shortcut **Ctrl + Alt + Y**.

System Information Parameter Definitions		
Field Name	**Values**	**Description**
Boot Version	nn.nn alpha-numeric	Indicates the version of the code that is used at the startup (booting) of the voip. The boot code version is independent of the software version.
Firmware Version	nn.nn.nn alpha-numeric	Indicates the version of the MultiVOIP firmware.
Configur-ation Version	nn.nn. nn.nn alpha-numeric	Indicates the version of the MultiVOIP configuration software.
Phone Book Version	nn.nn alpha-numeric	Indicates the version of the MultiVOIP phone book being used.
Mac Address	numeric	Denotes the number assigned as the voip unit's unique Ethernet address.
Up Time	days: hours: mm:ss	Indicates how long the voip has been running since its last booting.
Hardware ID	alpha-numeric	Indicates version of the MultiVOIP circuit board assembly being used.

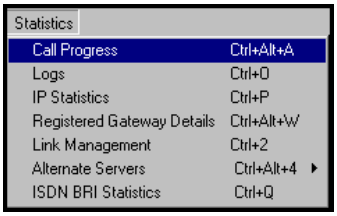
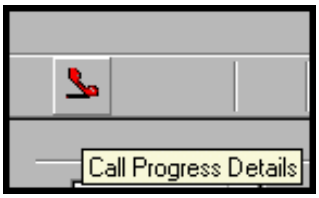
The frequency with which the System Information screen is updated is determined by a setting in the Logs screen



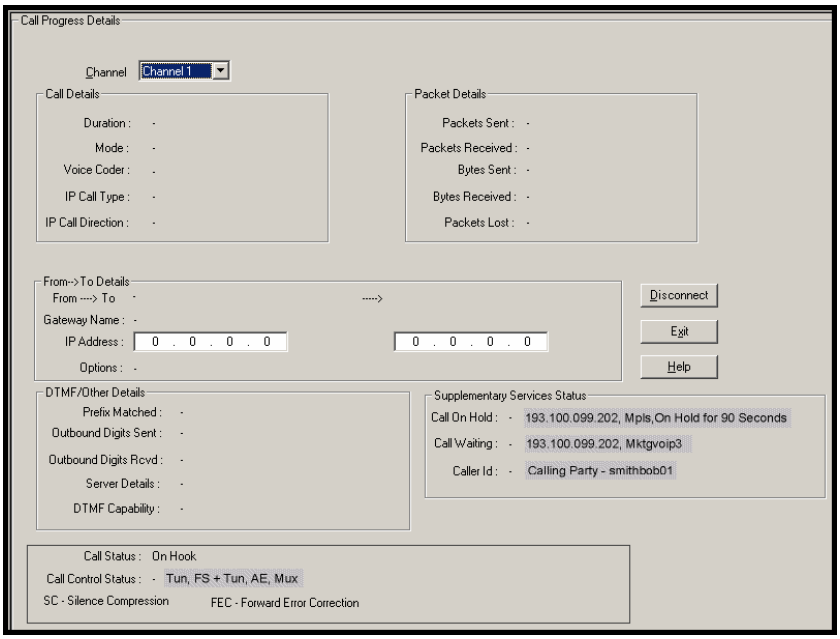
Statistics Screens

Ongoing operation of the MultiVOIP, whether it is in a MultiVOIP/PBX setting or MultiVOIP/telco-office setting, can be monitored for performance using the Statistics functions of the MultiVOIP software.

About Call Progress

<table>
<tr><th colspan="2">Accessing Call-Progress Statistics</th></tr>
<tr><td colspan="2">Channel Icons (Main Screen Lower Left)</td></tr>
<tr><td colspan="2">Channel icons are green when data traffic is present, red when idle.
 </td></tr>
<tr><td colspan="2">In the web GUI, call progress details can be viewed by clicking on an icon (one for each channel) arranged similarly on the web-browser screen.</td></tr>
<tr><td></td><td></td></tr>
<tr><td>Pulldown</td><td>Icon</td></tr>
<tr><td> 
</td><td> 
</td></tr>
<tr><td></td><td></td></tr>
<tr><td>Shortcut</td><td>Sidebar</td></tr>
<tr><td>Ctrl + Alt + A</td><td>
</td></tr>
</table>

The Call Progress Details Screen



Call Progress Details: Field Definitions		
Field Name	**Values**	**Description**
Channel	1-n	Number of data channel or time slot on which the call is carried. This is the channel for which call-progress details are being viewed.
Call Details		
Duration	Hours: Minutes: Seconds	The length of the call in hours, minutes, and seconds (hh:mm:ss).
Mode	Voice or FAX	Indicates whether the call being described was a voice call or a FAX call.
Voice Coder	G.723, G.729, G.711, etc.	The voice coder being used on this call.
IP Call Type	H.323, SIP, or SPP	Indicates the Call Signaling protocol used for the call (H.323, SIP, or SPP).
IP Call Direction	incoming, outgoing	Indicates whether the call in question is an incoming call or an outgoing call.

Call Progress Details: Field Definitions		
Field Name	**Values**	**Description**
Packet Details		
Packets Sent	integer value	The number of data packets sent over the IP network in the course of this call.
Packets Rcvd	integer value	The number of data packets received over the IP network in the course of this call.
Bytes Sent	integer value	The number of bytes of data sent over the IP network in the course of this call.
Bytes Rcvd	integer value	The number of bytes of data received over the IP network in the course of this call.
Packets Lost	integer value	The number of voice packets from this call that were lost after being received from the IP network.

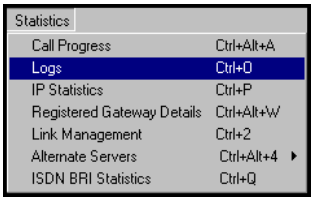
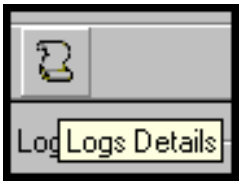
Call Progress Details: Field Definitions (cont'd)		
From – To Details		**Description**
Gateway Name (from)	alphanumeric string	Identifier for the VOIP gateway that handled the origination of this call.
IP Address (from)	x.x.x.x, where x has a range of 0 to 255	IP address from which the call was received.
Options	SC, FEC	Displays VOIP transmission options in use on the current call. These may include Forward Error Correction or Silence Compression.
Gateway Name (to)	alphanumeric string	Identifier for the VOIP gateway that handled the completion of this call.
IP Address (to)	x.x.x.x, where x has a range of 0 to 255	IP address to which the call was sent.
Options	SC, FEC	Displays VOIP transmission options in use on the current call. These may include Forward Error Correction or Silence Compression.

Call Progress Details: Field Definitions (cont'd)		
DTMF/Other Details		
Field Name	**Values**	**Description**
Prefix Matched	specified dialing digits	Displays the dialed digits that were matched to a phonebook entry.
Outbound Digits Sent	0-9, #, *	The digits transmitted by the MultiVOIP to the PBX/telco for this call.
Outbound Digits Received	0-9, #, *	Of the digits transmitted by the MultiVOIP to the PBX/telco for this call, these are the digits that were confirmed as being received.
Server Details	n.n.n.n (for n=0-255) and/or other server IP-related descriptions	The IP address (etc.) of the traffic control server (if any) being used (whether an H.323 gatekeeper, a SIP proxy, or an SPP registrar gateway) will be displayed here if the call is handled through that server.
DTMF Capability	inband, out of band Expressions differ slightly for different Call Signaling protocols (H.323, SIP, or SPP).	Indicates whether the DTMF dialing digits are carried "Inband" or "Out of Band." The corresponding field values differ for the 3 different voip protocols. For H.323, this field can display "Out of Band" or "Inband". For SIP it can display either "Out of Band RFC2833" or "Out of Band SIP INFO" to indicate the out-of-band condition or "Inband" to indicate the in-band condition. For SPP it can display "Out of Band RFC2833" or "Inband".

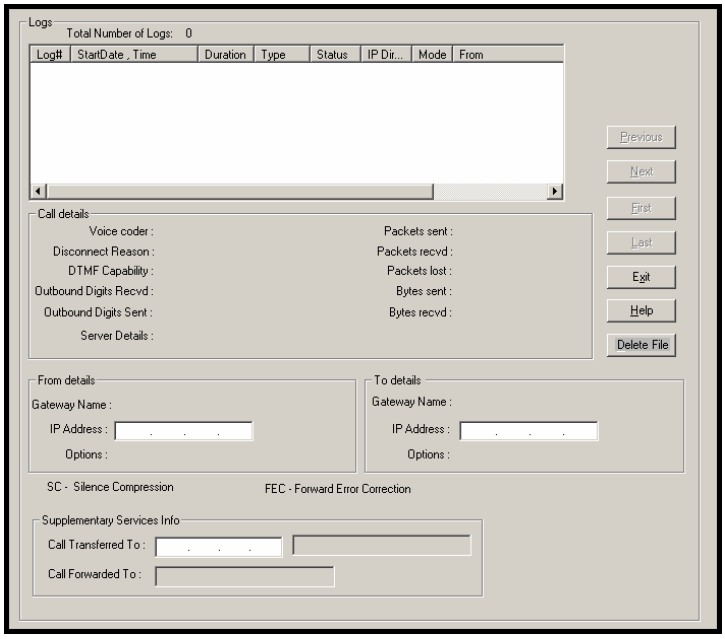
Call Progress Details: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Supplementary Services Status		
Call on Hold	alphanumeric	Describes held call by its IP address source, location/gateway identifier, and hold duration. Location/gateway identifiers comes from **Gateway Name** field in **Phone Book Configuration** screen of remote voip.
Call Waiting	alphanumeric	Describes waiting call by its IP address source, location/gateway identifier, and hold duration. Location/gateway identifiers comes from **Gateway Name** field in **Phone Book Configuration** screen of remote voip.
Caller ID	There are four values: "Calling Party + *identifier*"; "Alerting Party + *identifier*"; "Busy Party + *identifier*"; and "Connected Party + *identifier*"	This field shows the identifier and status of a remote voip (which has Call Name Identification enabled) with which this voip unit is currently engaged in some voip transmission. The status of the engagement (Connected, Alerting, Busy, or Calling) is followed by the identifier of a specific channel of a remote voip unit. This identifier comes from the "Caller Id" field in the **Supplementary Services** screen of the remote voip unit.

Call Progress Details: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Call Status fields		
Call Status	hangup, active	Shows condition of current call.
Call Control Status	Tun, FS + Tun, AE, Mux	Displays the H.323 version 4 features in use for the selected call. These include tunneling (Tun), Fast Start with tunneling (FS + Tun), Annex E multiplexed UDP call signaling transport (AE), and Q.931 Multiplexing (Mux). See **Phonebook Configuration Parameters** (in T1 or E1 chapters) for more on H.323v4 features.
Silence Compression	SC	"SC" stands for Silence Compression. With **Silence Compression** enabled, the MultiVOIP will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth that is being used by the voice channel.
Forward Error Correction	FEC	"FEC" stands for Forward Error Correction. **Forward Error Correction** enables some of the voice packets that were corrupted or lost to be recovered. FEC adds an additional 50% overhead to the total network bandwidth consumed by the voice channel. Default = Off

About Logs

Accessing "Statistics: Logs"	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
`Ctrl + O`	

The Logs Screen



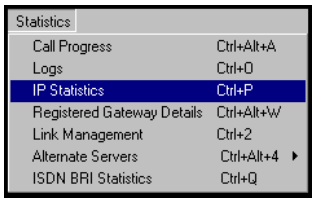
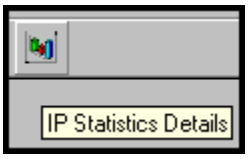
Logs Screen Details: Field Definitions		
Field Name	**Values**	**Description**
Log # column	1 or higher	All calls are assigned an event number in chronological order, with the most recent call having the highest event number.
Start Date,Time column	dd:mm:yyyy hh:mm:ss	The starting time of the call (event). The date is presented as a day expression of one or two digits, a month expression of one or two digits, and a four-digit year. This is followed by a time-of-day expression presented as a two-digit hour, a two-digit minute, and a two-digit seconds value. (statistics, logs) field
Duration column	hh:mm:ss	This describes how long the call (event) lasted in hours, minutes, and seconds.
Type	H.323, SIP, or SPP	Indicates the Call Signaling protocol used for the call (H.323, SIP, or SPP).
Status column	success or failure	Displays the status of the call, i.e., whether the call was completed successfully or not.
IP Direction	incoming, outgoing	Indicates whether the call is "incoming" or "outgoing" with respect to the gateway.
Mode column	voice or FAX	Indicates whether the (event) being described was a voice call or a FAX call.
From column	gateway name	Displays the name of the voice gateway that originates the call.
To column	gateway name	Displays the name of the voice gateway that completes the call.
Special Buttons		
Previous	--	Displays log entry before currently selected one.
Next	--	Displays log entry after currently selected one.
First	--	Displays first log entry
Last	--	Displays last log entry.
Delete File	--	Deletes selected log file.

Logs Screen Details: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Call Details		
Voice coder	G.723, G.729, G.711, etc.	The voice coder being used on this call.
Disconnect Reason	Values are "Normal" and "Local" disconnection.	Indicates whether the call was disconnected simply because the desired conversation was done or some other irregular cause occasioned disconnection (e.g., a technical error or failure).
DTMF Capability	inband, out of band Expressions differ slightly for different Call Signaling protocols (H.323, SIP, or SPP).	Indicates whether the DTMF dialing digits are carried "Inband" or "Out of Band." The corresponding field values differ for the 3 different voip protocols. For H.323, this field can display "Out of Band" or "Inband". For SIP it can display either "Out of Band RFC2833" or "Out of Band SIP INFO" to indicate the out-of-band condition or "Inband" to indicate the in-band condition. For SPP it can display "Out of Band RFC2833" or "Inband".
Outbound Digits Received	0-9, #, *	The DTMF dialing digits received by this gateway from the remote gateway presuming that DTMF is set to "Out of Band."
Outbound Digits Sent	0-9, #, *	The digits transmitted by the MultiVOIP to the PBX/telco for this call.
Server Details	a.b.c.d (standard IP octets; for a, b, c, d in the range 0 to 255)	The IP address of the traffic control server (if any) being used (whether an H.323 gatekeeper, a SIP proxy, or an SPP registrar gateway) will be displayed here if the call is handled through that server.

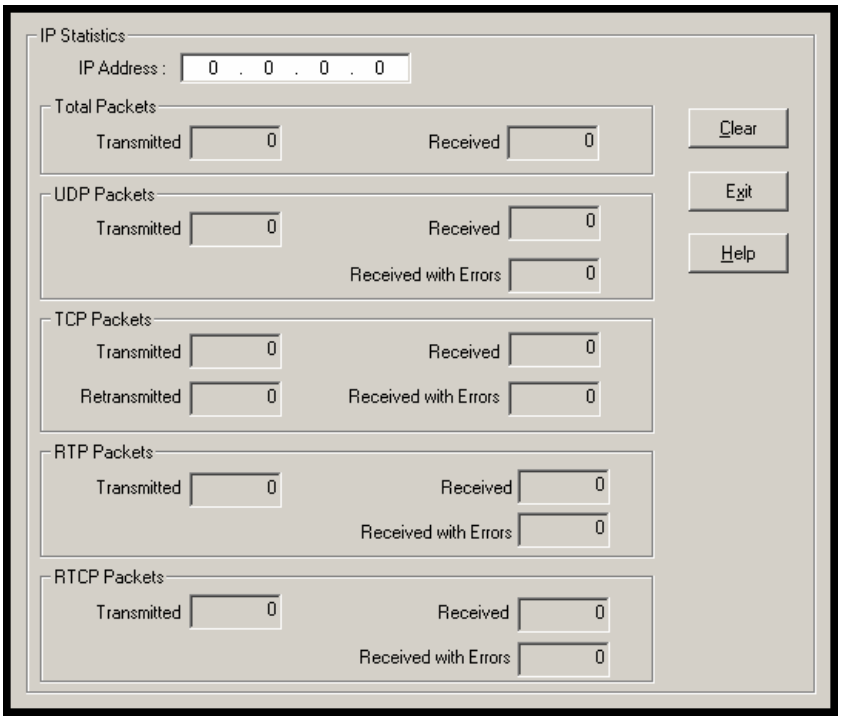
Logs Screen Details: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Call Details		
Packets sent	integer value	The number of data packets sent over the IP network in the course of this call.
Packets received	integer value	The number of data packets received over the IP network in the course of this call.
Packets loss (lost)	integer value	The number of voice packets from this call that were lost after being received from the IP network.
Bytes sent	integer value	The number of bytes of data sent over the IP network in the course of this call.
Bytes received	integer value	The number of bytes of data received over the IP network in the course of this call.
FROM Details		
Gateway Name	alphanumeric string	Identifier for the VOIP gateway that originated this call.
IP Address	x.x.x.x, where x has a range of 0 to 255	IP address of the VOIP gateway from which the call was received.
Options	FEC, SC	Displays VOIP transmission options used by the VOIP gateway originating the call. These may include Forward Error Correction or Silence Compression.
TO Details		
Gateway Name	alphanumeric string	Identifier for the VOIP gateway that completed (terminated) this call.
IP Address	x.x.x.x, where x has a range of 0 to 255	IP address of the VOIP gateway at which the call was completed (terminated).
Options		Displays VOIP transmission options used by the VOIP gateway terminating the call. These may include Forward Error Correction or Silence Compression.

Logs Screen Details: Field Definitions (cont'd)		
Supplementary Services Info		
Call Transferred To	phone number string	Number of party called in transfer.
Call Forwarded To	phone number string	Number of party called in forwarding.

About IP Statistics

Accessing IP Statistics	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
`Ctrl + P`	

IP Statistics Screen



UDP Versus TCP

User Datagram Protocol versus Transmission Control Protocol. UDP provides unguaranteed, connectionless transmission of data across an IP network. By contrast, TCP provides reliable, connection-oriented transmission of data.

Both TCP and UDP split data into packets called "datagrams." However, TCP includes extra headers in the datagram to enable retransmission of lost packets and reassembly of packets into their correct order if they arrive out of order. UDP does not provide this. Lost UDP packets are unretrievable; that is, out-of-order UDP packets cannot be reconstituted in their proper order..

Despite these obvious disadvantages, UDP packets can be transmitted much faster than TCP packets -- as much as three times faster. In certain applications, like audio and video data transmission, the need for high speed outweighs the need for verified data integrity. Sound or pictures often remain intelligible despite a certain amount of lost or disordered data packets (which appear as static).

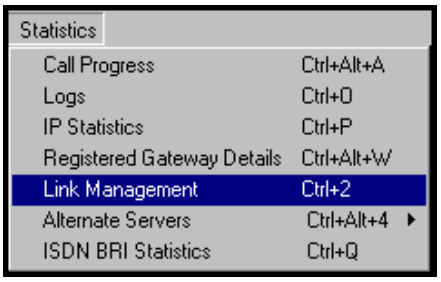
IP Statistics: Field Definitions		
Field Name	**Values**	**Description**
IP Address	n.n.n.n 0 - 255	IP address of the MultiVOIP. For an IP address to be displayed here, the MultiVOIP must have DHCP enabled. Its IP address, in such a case, is assigned by the DHCP server.
"Clear" button	--	Clears packet tallies from memory.
Total Packets		Sum of data packets of all types.
Transmitted	integer value	Total number of packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Total number of packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Total number of error-laden packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.

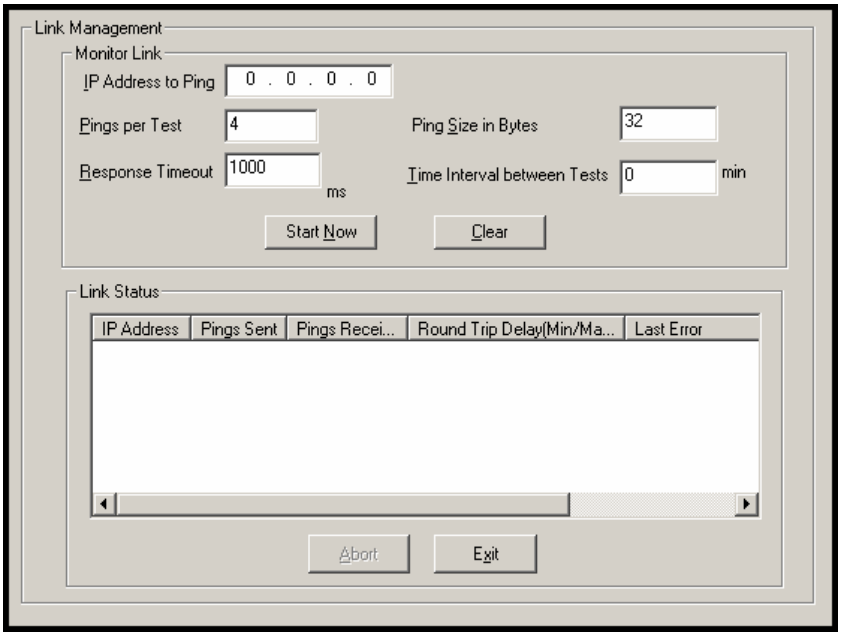
IP Statistics: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
UDP Packets		User Datagram Protocol packets.
Transmitted	integer value	Number of UDP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of UDP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden UDP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
TCP Packets		Transmission Control Protocol packets.
Transmitted	integer value	Number of TCP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of TCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden TCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.

IP Statistics: Field Definitions (cont'd)		
RTP Packets		Voice signals are transmitted in Realtime Transport Protocol packets. RTP packets are a type or subset of UDP packets.
Transmitted	integer value	Number of RTP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of RTP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden RTP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
RTCP Packets		Realtime Transport Control Protocol packets convey control information to assist in the transmission of RTP (voice) packets. RTCP packets are a type or subset of UDP packets.
Transmitted	integer value	Number of RTCP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of RTCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden RTCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.

About Link Management

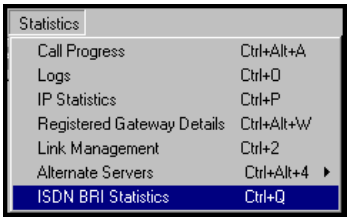
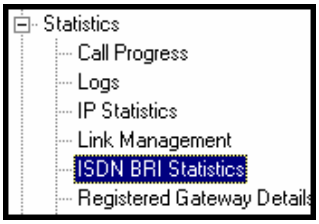
The Link Management screen is essentially an automated utility for pinging endpoints on your voip network. This utility generates pings of variable sizes at variable intervals and records the response to the pings.

<table>
<tr><th colspan="2">Accessing Link Management</th></tr>
<tr><th colspan="2">Pulldown</th></tr>
<tr><td colspan="2"> 
</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut // Icon</th><th>Sidebar</th></tr>
<tr><td>Ctrl + 2 // none</td><td>
</td></tr>
</table>

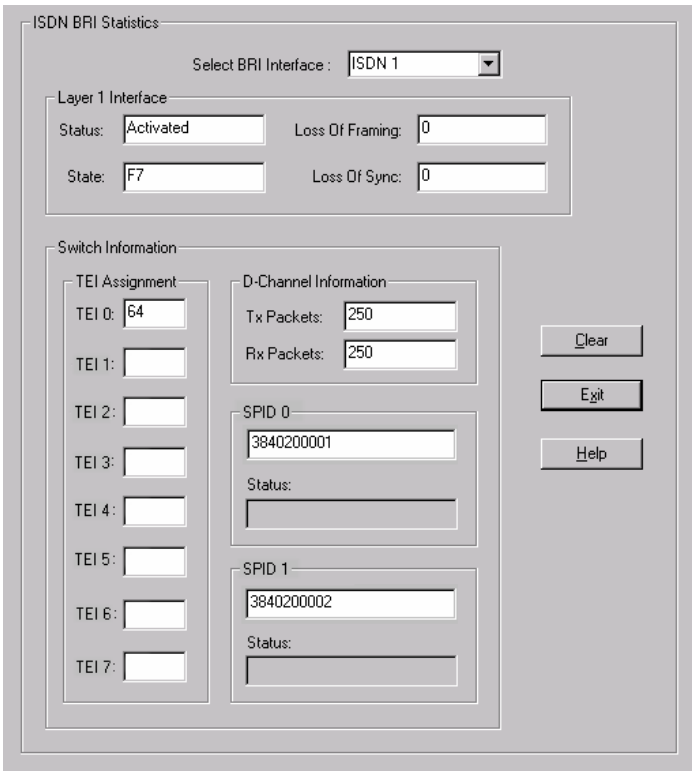

Link Management
Monitor Link
IP Address to Ping
0 . 0 . 0 . 0
Pings per Test
4
Ping Size in Bytes
32
Response Timeout
1000
ms
Time Interval between Tests
0
min
Start Now
Clear
Link Status
IP Address
Pings Sent
Pings Recei...
Round Trip Delay(Min/Ma...
Last Error
Abort
Exit

Link Management screen Field Definitions		
Field Name	**Values**	**Description**
Monitor Link fields		
IP Address to Ping	a.b.c.d 0-255	This is the IP address of the target endpoint to be pinged.
Pings per Test	1-999	This field determines how many pings will be generated by the Start Now command.
Response Timeout	500 – 5000 milliseconds	The duration after which a ping will be considered to have failed.
Ping Size in Bytes	32 – 128 bytes	This field determines how long or large the ping will be.
Timer Interval between Pings	0 or 30 – 6000 minutes	This field determines how long of a wait there is between one ping and the next.
Start Now command button	--	Initiates pinging.
Clear command button	--	Erases ping parameters in Monitor Link field group and restores default values.
Link Status Parameters		These fields summarize the results of pinging.
IP Address column	a.b.c.d 0-255	Target of ping.
No. of Pings Sent	as listed	Number of pings sent to target endpoint.
No. of Pings Received	as listed	Number of pings received by target endpoint.
Round Trip Delay (Min/Max/ Avg)	as listed, in milliseconds	Displays how long it took from time ping was sent to time ping response was received.
Last Error	as listed	Indicates when last data error occurred.

About ISDN BRI Statistics

Accessing BRI Statistics	
Pulldown	**Sidebar**
	
Shortcut	**Icon**
`Ctrl + Q`	--

ISDN BRI Statistics Screen



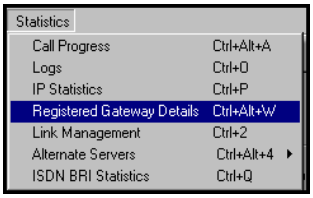
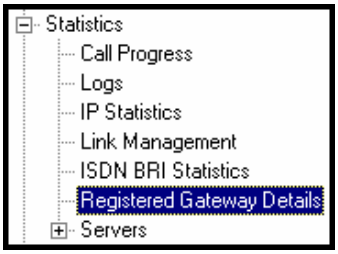
ISDN BRI Statistics: Field Definitions		
Field Name	**Values**	**Description**
Select BRI Interface	ISDNn For n=1-2 (410ST) For n-1-4 (810ST)	In this field, you can choose the ISDN port for which you want to view the status. The 410ST has two ISDN –BRI ports (or "interfaces"); the 810ST has four ISDN-BRI ports (or "interfaces"). Each interface has two channels.
Layer 1 Interface		
Status	inactive (F1), sensing (F2), deactivated (F3), awaiting signal (F4), identifying input (F5), synchronized (F6), activated (F7), lost framing (F8), deactive (G1), pending activation (G2), active (G3), pending deactivation (G4)	Shows the current Layer 1 status of the ISDN connection. Each status description (inactive, sensing, etc.) corresponds to a particular "state" label (F1-F8 and G1-G4).
State	F1-F8 (for Terminal mode ports), G1-G4 (for Network mode ports)	Shows the I.430 state name for Layer 1. An "F" state name indicates this port is in Terminal mode (F1-F8), as set in the **ISDN BRI Parameters** screen. A "G" state name indicates that this port is in Network mode (G1-G4), as set in the **ISDN BRI Parameters** screen.
Loss Of Framing	integer	Shows the number of lost-framing events on the ISDN physical layer.
Loss of Sync	integer	Shows the number of lost-synchronization events on the ISDN physical layer.

ISDN BRI Statistics: Field Definitions (continued)		
Field Name	**Values**	**Description**
Switch Information: TEI Assignment		
TEI 0 through TEI 7	0-63 (point-to-point assignments) 64-126 (automatic assignments)	Displays the value for each TEI assigned to the BRI port. The TEI (Terminal Endpoint Identifier) uniquely identifies each device connected to the ISDN physical layer.
Switch Information: D-Channel Information		
Tx Packets	0 to 4294967295	Shows the number of packets transmitted on the channel. When the value exceeds 4294967295 packets, it will reset to zero and continue counting.
Rx Packets	0 to 4294967295	Shows the number of packets received on the channel. When the value exceeds 4294967295 packets, it will reset to zero and continue counting.
Switch Information: SPID 0		
(SPID 0 *number*)	numeric, 3 to 20 digits	A SPID (Service Profile Identifier) is assigned by the ISDN provider and pertains to one channel of the BRI interface (port), in this case channel 0. The SPID identifies an ISDN terminal uniquely. The SPID associates a set of services (features) with the terminal. (In Terminal mode the provider is a telco or PBX. In Network mode MultiVOIP is the provider.) A SPID is only used when the "Country" field is set to "USA" in the **ISDN BRI Parameters** screen.
Status	Not Checked, Correct, Incorrect	Indicates whether SPID0 is correct, incorrect, or not being checked.

ISDN BRI Statistics: Field Definitions (continued)		
Field Name	**Values**	**Description**
Switch Information: SPID 1		
(SPID 1 *number*)	numeric	SPID for channel 1 of the BRI interface. Otherwise, same as SPID0 description above.
Status	Not Checked, Correct, Incorrect	Indicates whether SPID1 is correct, incorrect, or not being checked.
"Clear" button		Clears (sets to zero) all **ISDN BRI Statistics** fields with numeric tally values (these are Loss of Framing, Loss of Sync, Tx Packets, Rx Packets).

About Registered Gateway Details

The Registered Gateway Details screen presents a real-time display of the special operating parameters of the Single Port Protocol (SPP). These are configured in the **Call Signaling** screen and in the **Add/Edit Outbound PhoneBook** screen.

<table>
<tr><th colspan="2">Accessing Registered Gateway Details</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td> 
</td><td>--</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>Ctrl + Alt + W</td><td> 
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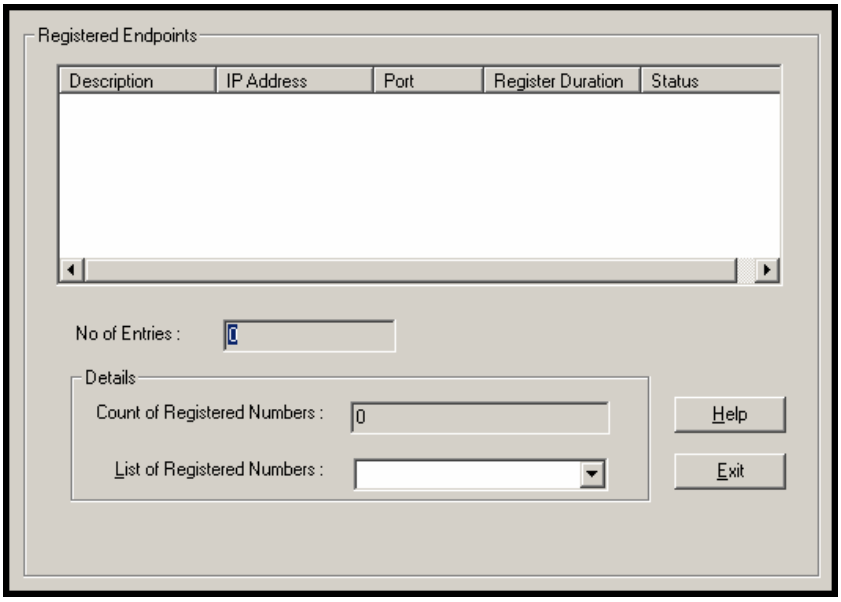
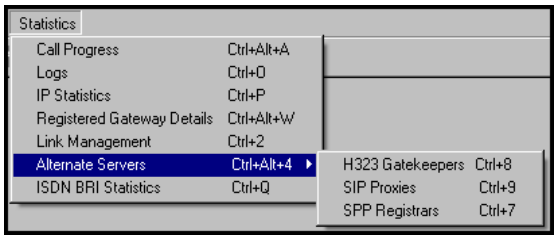
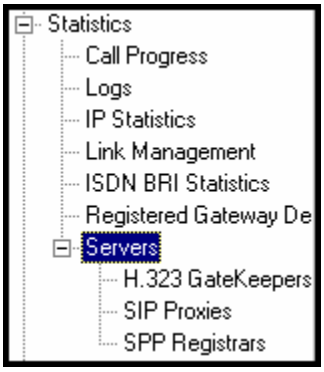

Registered Endpoints
Description
IP Address
Port
Register Duration
Status
No of Entries :
0
Details
Count of Registered Numbers :
0
List of Registered Numbers :
Help
Exit

Registered Gateway Details: Field Definitions		
Field Name	**Values**	**Description**
Column Headings		
Description	alphanumeric	This is a descriptor for a particular voip gateway unit. This descriptor should generally identify the physical location of the unit (e.g., city, building, etc.) and perhaps even its location in an equipment rack.
IP Address	n.n.n.n, for n = 0-255	The RAS address for the gateway.
Port		Port by which the gateway exchanges H.225 RAS messages with the gatekeeper. .
Register Duration		The time remaining in seconds before the TimeToLive timer expires. If the gateway fails to reregister within this time, the endpoint is unregistered.
Status		The current status of the gateway, either registered or unregistered.
No. of Entries		The number of gateways currently registered to the Registrar. This includes all SPP clients registered and the Registrar itself.
Details		
Count of Registered Numbers		If a registered gateway is selected (by clicking on it in the screen), The "Count of Registered Numbers" will indicate the number of registered phone numbers for the selected gateway. When a client registers, all of its inbound phonebook's phone numbers become registered.
List of Registered Numbers		Lists all of the registered phone numbers for the selected gateway.

About Alternate Server Statistics

<table>
<tr><th colspan="2">Accessing Alternate Server Statistics</th></tr>
<tr><th colspan="2">Pulldown</th></tr>
<tr><td colspan="2"> 
</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>Ctrl + Alt + 4</td><td> 
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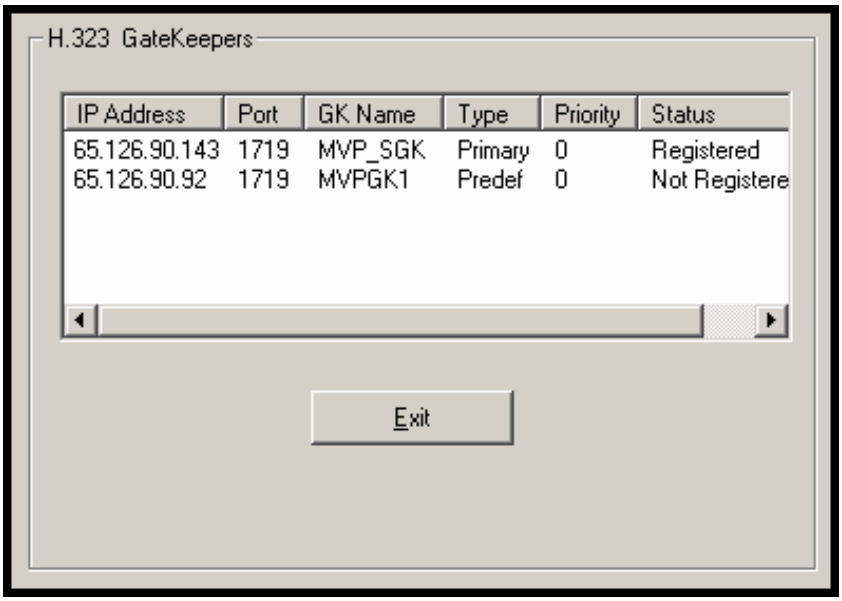
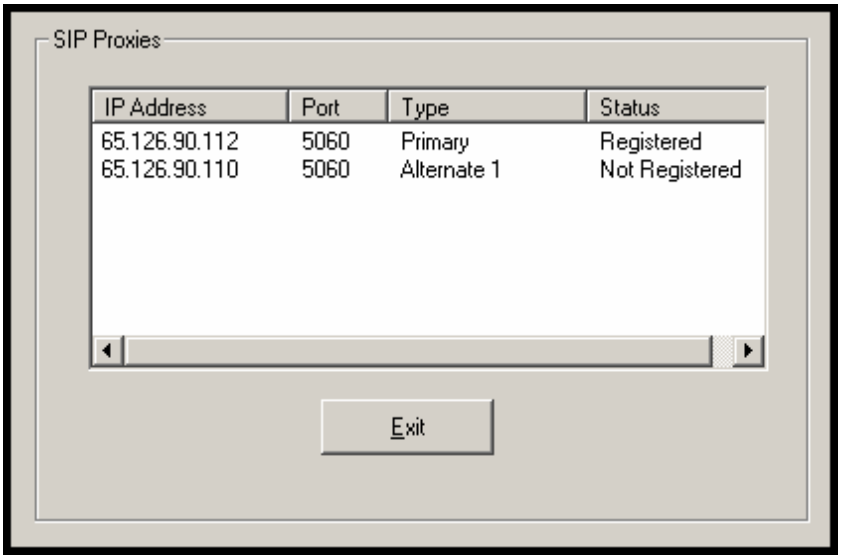
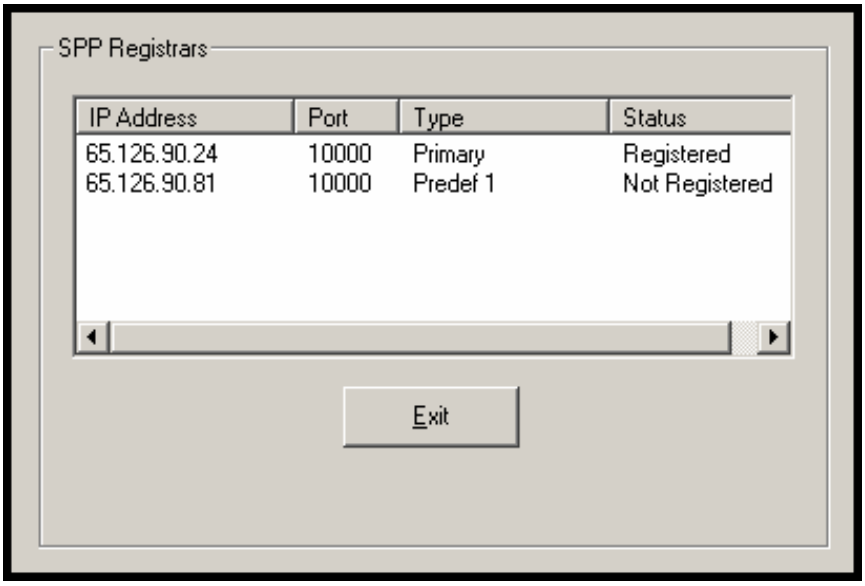


H.323 Gatekeepers (Statistics, Servers): Field Definitions		
Field Name	**Values**	**Description**
Column Headings		
IP Address	n.n.n.n, for n = 0-255	The IP address of the gatekeeper.
Port		TDMA time slot used for communication between MultiVOIP unit and the gatekeeper that serves it.
GK Name	alpha-numeric string	Identifier for gatekeeper.
Type	Primary, Predefined	This field describes the type of gateway as which the MultiVOIP is defined with respect to the gatekeeper.
Priority		Priority refers to … .
Status	registered, not registered	The current status of the gateway, either registered or unregistered.



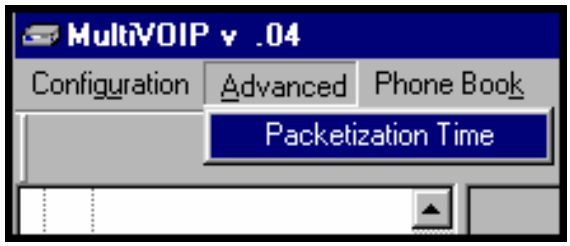
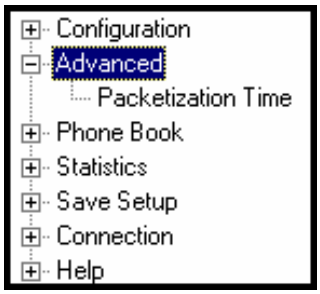
SIP Proxies (Statistics, Servers): Field Definitions		
Field Name	**Values**	**Description**
Column Headings		
IP Address	n.n.n.n, for n = 0-255	The IP address of the SIP proxy by which the MultiVOIP is governed.
Port		TDMA time slot used for communication between MultiVOIP unit and the SIP Proxy that governs it.
Type	Primary, Alternate	This field describes the type of gateway as which the MultiVOIP is defined with respect to the gatekeeper.
Status	registered, not registered	The current status of the MultiVOIP gateway with respect to the SIP proxy, either registered or unregistered.



SPP Registrars (Statistics, Servers): Field Definitions		
Field Name	**Values**	**Description**
Column Headings		
IP Address	n.n.n.n, for n = 0-255	The IP address of the gatekeeper.
Port		TDMA time slot used for communication between MultiVOIP unit and the gatekeeper that serves it.
Type	Primary, Predefined	This field describes the type of gateway as which the MultiVOIP is defined with respect to the gatekeeper.
Status	registered, not registered	The current status of the gateway, either registered or unregistered.

About Packetization Time

You can use the **Packetization Time** screen to specify definite packetization rates for coders selected in the Voice/FAX Parameters screen (in the "Coder Options" group of fields). The Packetization Time screen is accessible under the "Advanced" options entry in the sidebar list of the main voip software screen. In dealing with RTP parameters, the Packetization Time screen is closely related to both Voice/FAX Parameters and to IP Statistics. It is located in the "Advanced" group for ease of use.

<table>
<tr><th colspan="2">Accessing Packetization Time</th></tr>
<tr><th colspan="2">Pulldown</th></tr>
<tr><td colspan="2"> 
</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut/Icon</th><th>Sidebar</th></tr>
<tr><td>none/none</td><td> 
</td></tr>
</table>

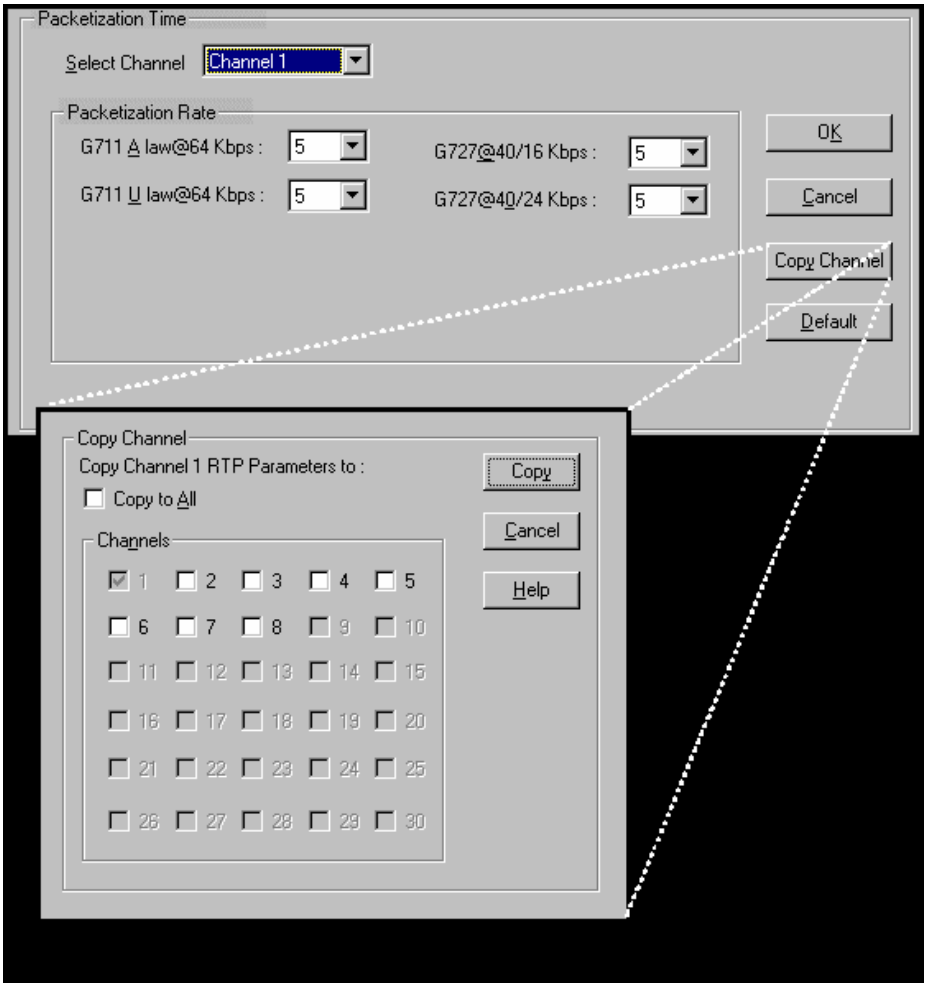
Packetization Time Screen

Packetization rates can be set separately for each channel.

The table below presents the ranges and increments for packetization rates.

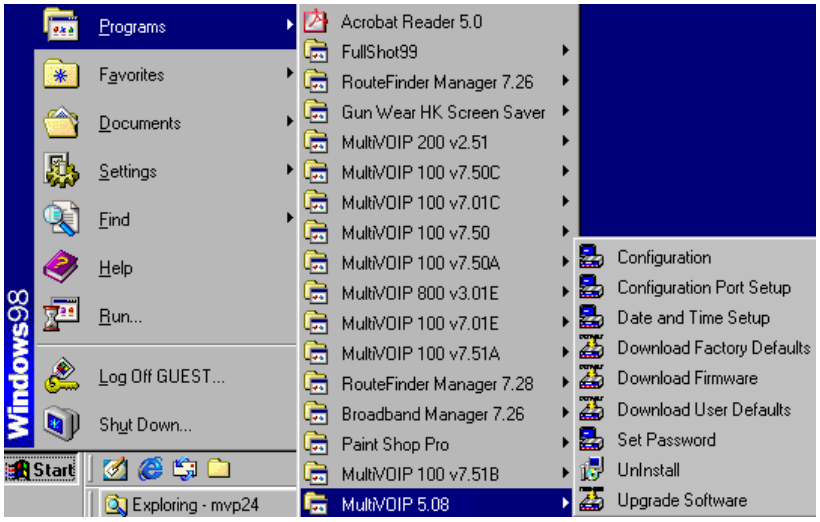
Packetization Ranges and Increments			
Coder Types	Range (in Kbps); {default value}		Increments (in Kbps)
G711, G726, G727	5-120	{5}	5
G723	30-120	{30}	30
G729	10-120	{10}	10
Netcoder	20-120	{20}	20

Once the packetization rate has been set for one channel, it can be copied into other channels.



MultiVoip Program Menu Items

After the MultiVoip program is installed on the PC, it can be launched from the **Programs** group of the Windows **Start** menu (**Start** | **Programs** | **MultiVOIP** ____ | …). In this section, we describe the software functions available on this menu.



Several basic software functions are accessible from the MultiVoip software menu, as shown below.

MultiVOIP Program Menu	
Menu Selection	**Description**
Configuration	Select this to enter the Configuration program where values for IP, telephony, and other parameters are set.
Configuration Port Setup	Select this to access the **COM Port Setup** screen of the MultiVOIP Configuration program.
Date and Time Setup	Select this for access to set calendar/clock used for data logging.

MultiVOIP Program Menu (cont'd)	
Menu Selection	**Description**
Download Factory Defaults	Select this to return the configuration parameters to the original factory values.
Download Firmware	Select this to download new versions of firmware as enhancements become available.
Download User Defaults	To be used after a full set of parameter values, values specified by the user, have been saved (using Save Setup). This command loads the saved user defaults into the MultiVOIP.
Set Password	Select this to create a password for access to the MultiVOIP software programs (**Program** group commands, Windows GUI, web browser GUI, & FTP server). Only the FTP Server function *requires* a password for access. The FTP Server function also requires that a username be established along with the password.
Uninstall	Select this to uninstall the MultiVOIP software (most, but not all components are removed from computer when this command is invoked).
Upgrade Software	Loads firmware (including H.323 stack) and factory default settings from the controller PC to the MultiVOIP unit.

"Downloading" here refers to transferring program files from the PC to the nonvolatile "flash" memory of the MultiVOIP. Such transfers are made via the PC's serial port. This can be understood as a "download" from the perspective of the MultiVOIP unit.

When new versions of the MultiVoip software become available, they will be posted on MultiTech's web or FTP sites. Although transferring updated program files from the MultiTech web/FTP site to the user's PC can generally be considered a download (from the perspective of the PC), this type of download cannot be initiated from the MultiVoip software's Program menu command set.

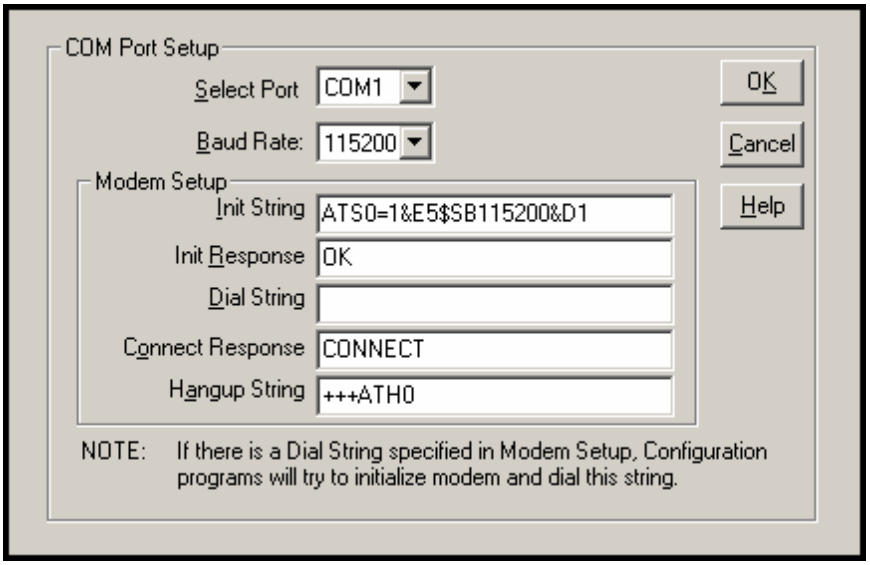
Generally, updated firmware must be downloaded from the MultiTech web/FTP site to the PC before it can be loaded from the PC to the MultiVOIP.

Configuration Option

The "Configuration" option in the MultiVOIP Program menu launches the MultiVOIP Configuration software program.

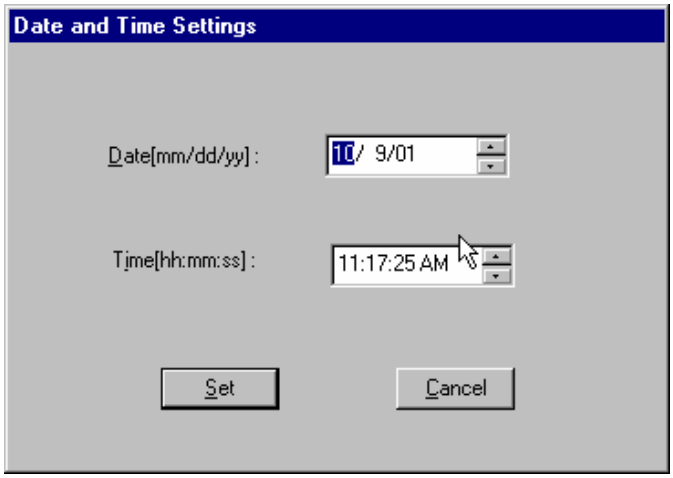
Configuration Port Setup

The Configuration Port Setup option in the MultiVOIP Program menu brings up the **COM Port Setup** screen of the MultiVOIP configuration software.



Date and Time Setup

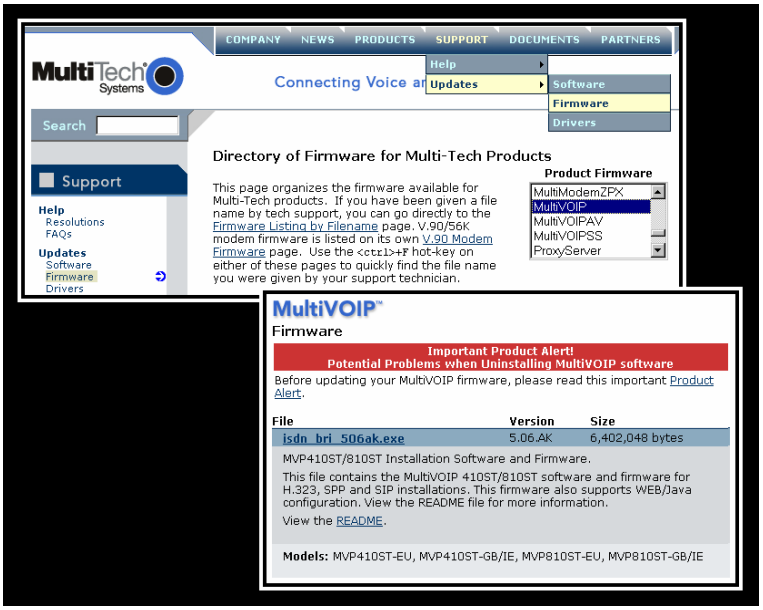
The dialog box below allows you to set the time and date indicators of the MultiVOIP system.



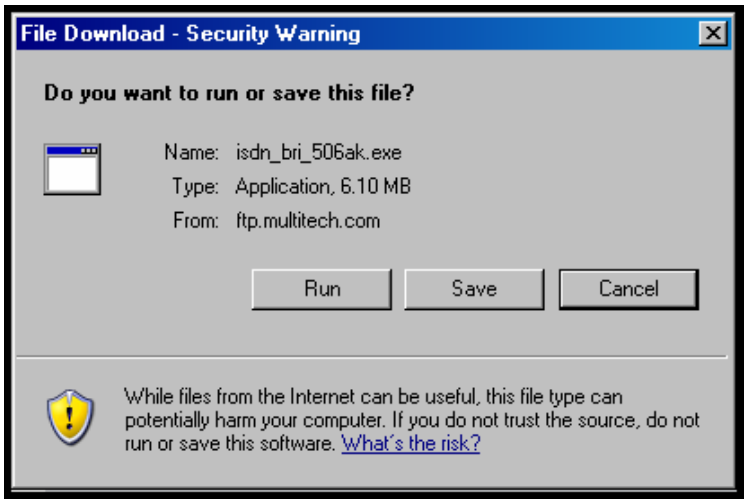
Obtaining Updated Firmware

Generally, updated firmware must be downloaded from the MultiTech web/FTP site to the user's PC before it can be downloaded from that PC to the MultiVOIP.

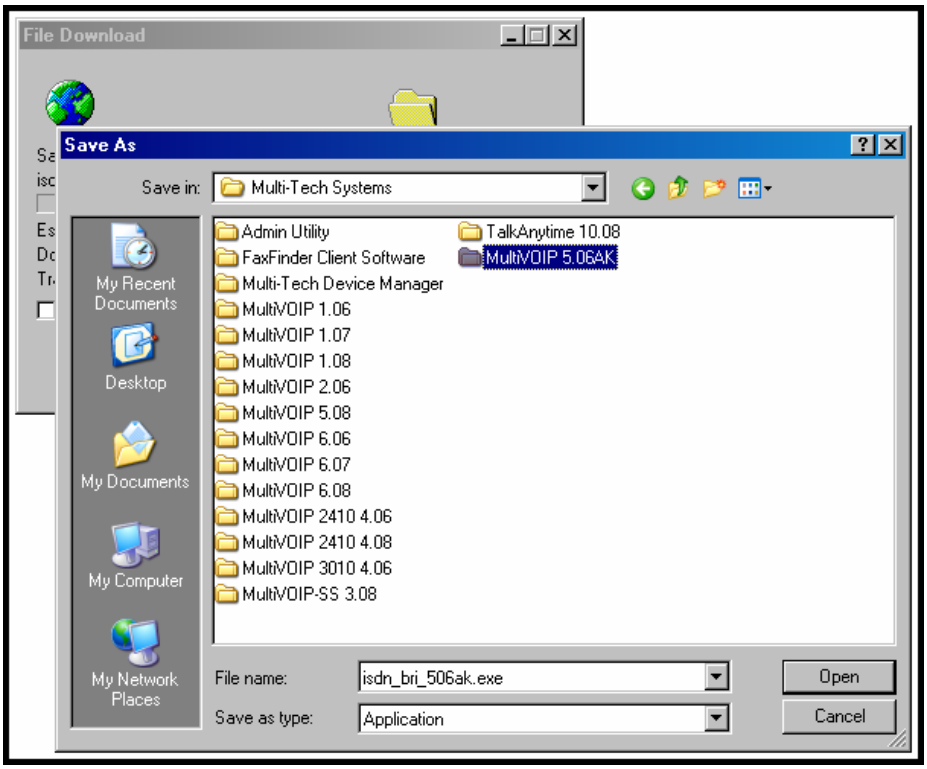
Note that the structure of the MultiTech web/FTP site may change without notice. However, firmware updates can generally be found using standard web techniques. For example, you can access updated firmware by doing a search or by clicking on **Support**. The MultiTech web site may also contain special notices or alerts that are not included in manuals.



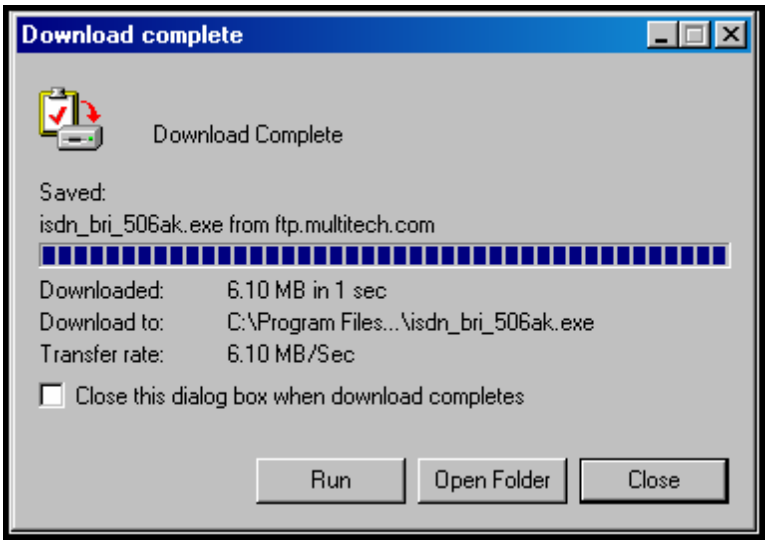
Once the updated firmware has been located, it can be downloaded from the web/ftp site using commands common to web/ftp downloading.


File Download - Security Warning
Do you want to run or save this file?
Name: isdn_bri_506ak.exe
Type: Application, 6.10 MB
From: ftp.multitech.com
Run
Save
Cancel
While files from the Internet can be useful, this file type can potentially harm your computer. If you do not trust the source, do not run or save this software. What's the risk?

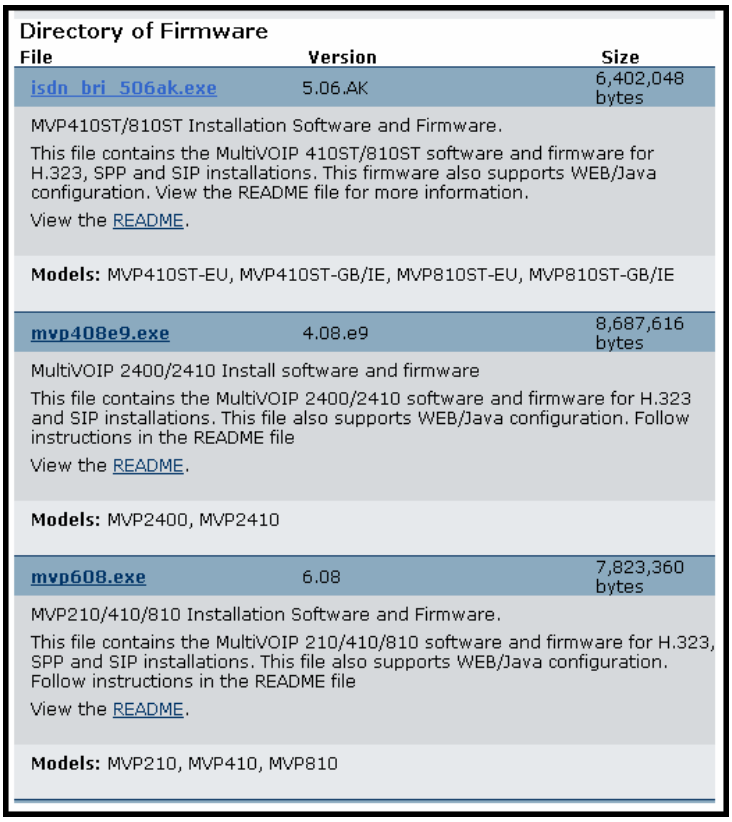
As with other downloads, you will want to put it in a fitting directory.



The progress of the file transfer will be shown until, finally, the download is complete.



The MultiTech web site also lists firmware by filename in its Directory of Firmware.



Directory of Firmware

File	Version	Size
isdn_bri_506ak.exe	5.06.AK	6,402,048 bytes

MVP410ST/810ST Installation Software and Firmware.

This file contains the MultiVOIP 410ST/810ST software and firmware for H.323, SPP and SIP installations. This firmware also supports WEB/Java configuration. View the README file for more information.

View the README.

Models: MVP410ST-EU, MVP410ST-GB/IE, MVP810ST-EU, MVP810ST-GB/IE

mvp408e9.exe	4.08.e9	8,687,616 bytes

MultiVOIP 2400/2410 Install software and firmware

This file contains the MultiVOIP 2400/2410 software and firmware for H.323 and SIP installations. This file also supports WEB/Java configuration. Follow instructions in the README file

View the README.

Models: MVP2400, MVP2410

mvp608.exe	6.08	7,823,360 bytes

MVP210/410/810 Installation Software and Firmware.

This file contains the MultiVOIP 210/410/810 software and firmware for H.323, SPP and SIP installations. This file also supports WEB/Java configuration. Follow instructions in the README file

View the README.

Models: MVP210, MVP410, MVP810

Implementing a Software Upgrade

MultiVOIP software can be upgraded locally using a single command at the MultiVOIP Windows GUI, namely **Upgrade Software**. This command downloads firmware (including the H.323 stack), and factory default settings from the controller PC to the MultiVOIP unit.

When using the MultiVOIP Windows GUI, firmware and factory default settings can also be transferred from controller PC to MultiVOIP piecemeal using separate commands.

When using the MultiVOIP web browser GUI to control/configure the voip remotely, upgrading of software must be done on a piecemeal basis using the FTP Server function of the MultiVOIP unit.

When performing a piecemeal software upgrade (whether from the Windows GUI or web browser GUI), follow these steps in order:

1. Identify Current Firmware Version
2. Download Firmware
3. Download Factory Defaults

When upgrading firmware, the software commands "Download Firmware," and "Download Factory Defaults" must be implemented in order, else the upgrade is incomplete.

Identifying Current Firmware Version

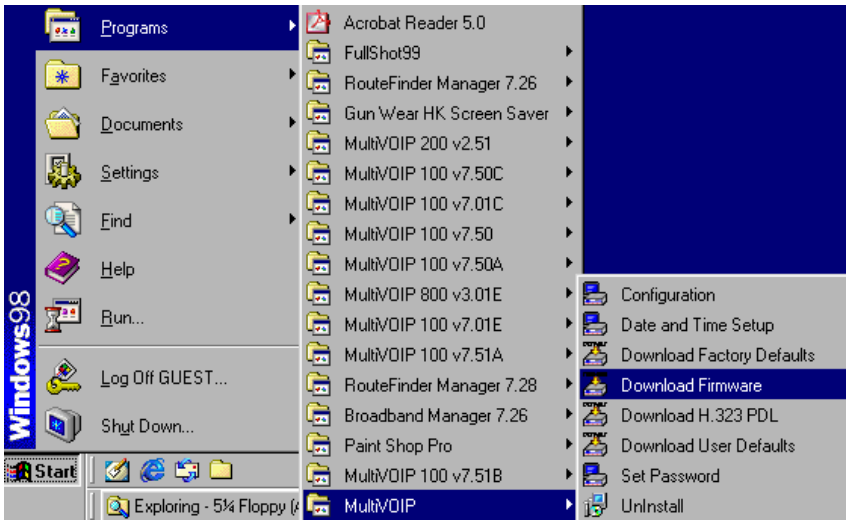
Before implementing a MultiVOIP firmware upgrade, be sure to verify the firmware version currently loaded on it. The firmware version appears in the MultiVoip Program menu. Go to **Start** | **Programs** | **MultiVOIP** ____ **x.xx**. The final expression, x.xx, is the firmware version number. In the illustration below, the firmware version is 4.00a, made for the E1 MultiVOIP (MVP3010).

When a new firmware version is installed, the MultiVOIP software can be upgraded in one step using the **Upgrade Software** command, or piecemeal using the **Download Firmware** command and the **Download Factory Defaults** command.

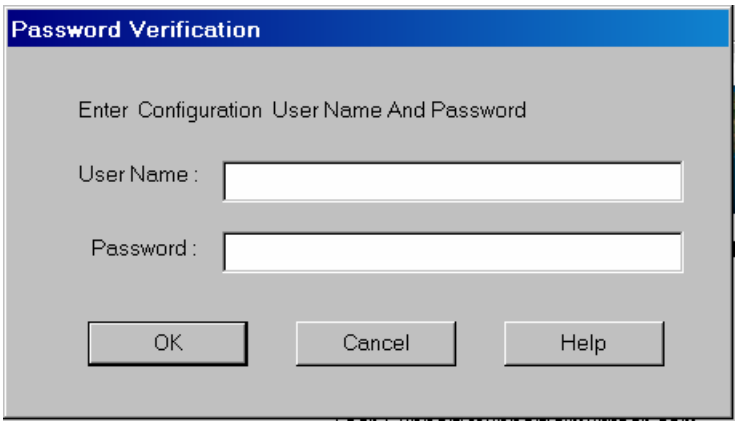
Download Firmware transfers the firmware (including the H.323 protocol stack) in the PC's MultiVOIP directory into the nonvolatile flash memory of the MultiVOIP.
Download Factory Defaults sets all configuration parameters to the standard default values that are loaded at the MultiTech factory.
Upgrade Software both implements the **Download Firmware** command and lets the user choose to retain the current configuration or to download factory default values.

Downloading Firmware

1. The MultiVoip Configuration program must be off when invoking the **Download Firmware** command. If it is on, the command will not work.
2. To invoke the Download Factory Defaults command, go to **Start** | **Programs** | **MVP____ x.xx** | **Download Firmware**.

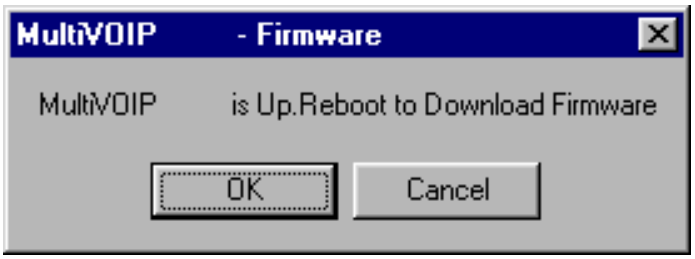


3. If a password has been established, the **Password Verification** screen will appear.



Type in the username and password and click **OK**.

4. The **MultiVOIP ___- Firmware** screen appears saying "MultiVOIP [*model number*] is up. Reboot to Download Firmware?"

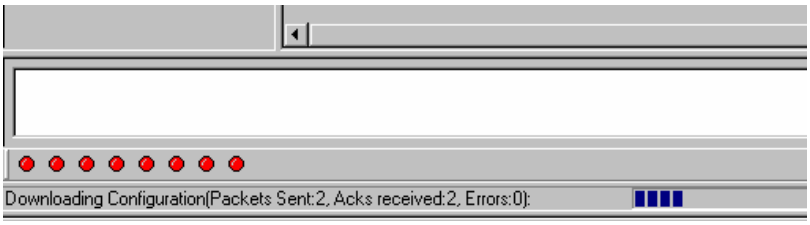


Click **OK** to download the firmware.

The "Boot" LED on the MultiVOIP will light up and remain lit during the file transfer process.

5. The program will locate the firmware ".bin" file in the MultiVOIP directory. Highlight the correct (newest) ".bin" file and click **Open**.

6. Progress bars will appear at the bottom of the screen during the file transfer.



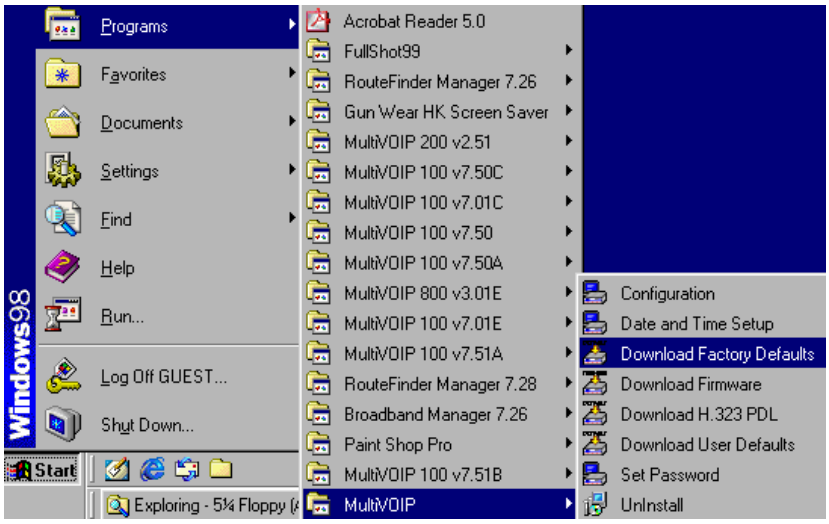
The MultiVOIP's "Boot" LED will turn off at the end of the transfer.

7. The **Download Firmware** procedure is complete.

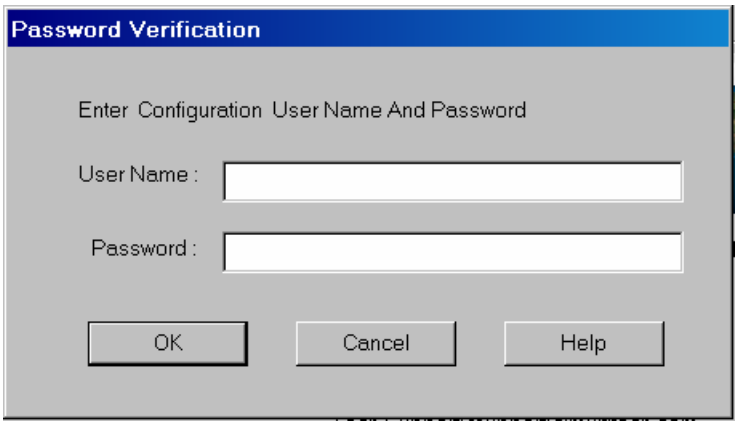
Downloading Factory Defaults

1. The MultiVoip Configuration program must be off when invoking the **Download Factory Defaults** command. If it is on, the command will not work.

2. To invoke the **Download Factory Defaults** command, go to **Start** | **Programs** | **MVP____ x.xx** | **Download Factory Defaults**.

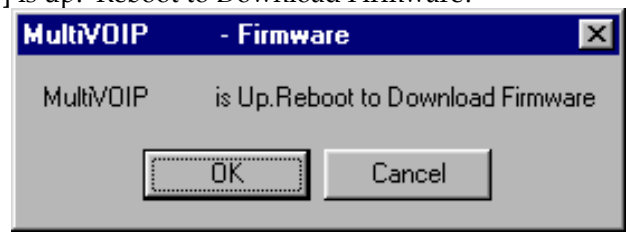


3. If a password has been established, the **Password Verification** screen will appear.



Type in the username and password and click **OK**.

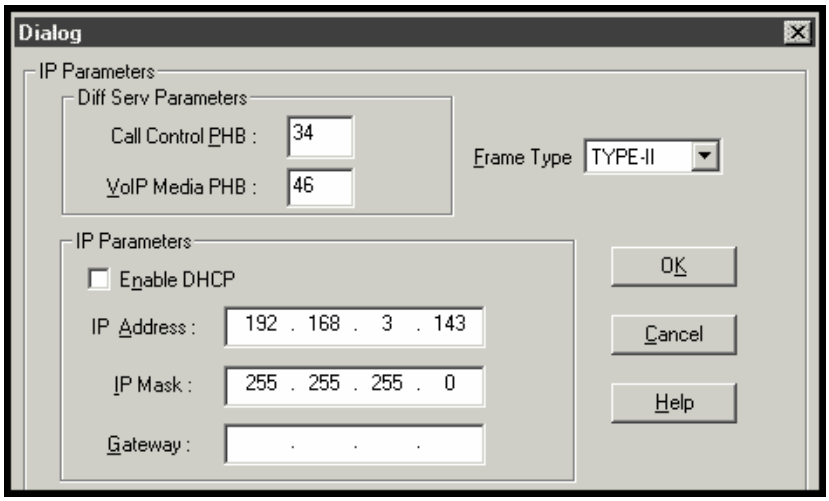
4. The **MVP____- Firmware** screen appears saying "MultiVOIP [*model number*] is up. Reboot to Download Firmware?"



Click **OK** to download the factory defaults.

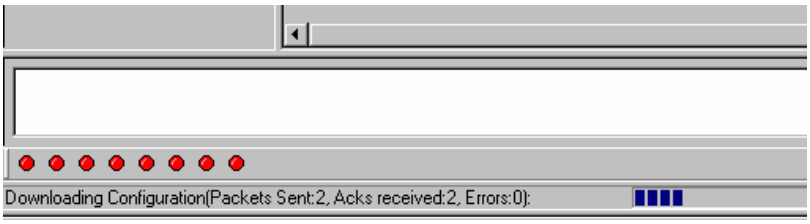
The "Boot" LED on the MultiVOIP will light up and remain lit during the file transfer process.

5. After the PC gets a response from the MultiVOIP, the **Dialog – IP Parameters** screen will appear.



The user should verify that the correct IP parameter values are listed on the screen and revise them if necessary. Then click **OK**.

6. Progress bars will appear at the bottom of the screen during the file transfer.

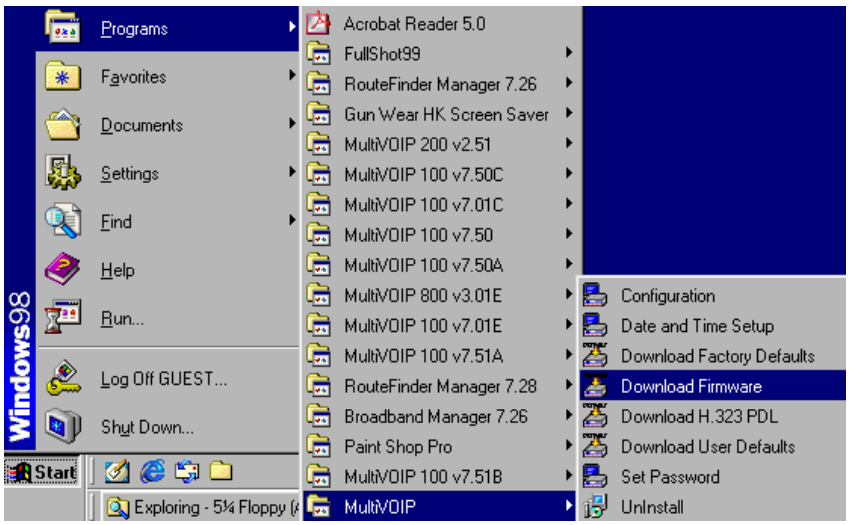


The MultiVOIP's "Boot" LED will turn off at the end of the transfer.

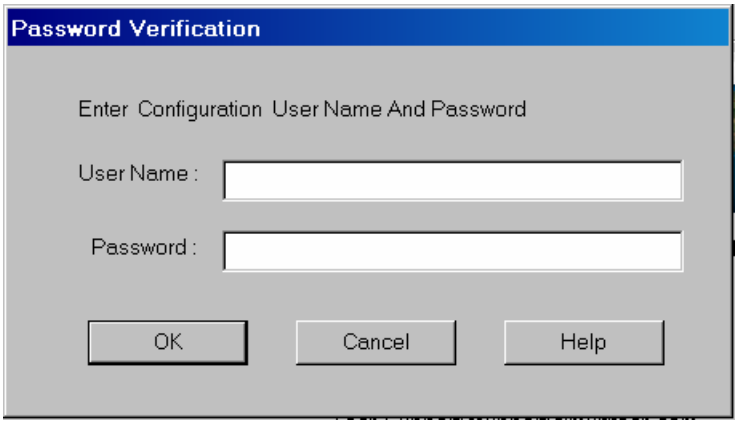
7. The **Download Factory Defaults** procedure is complete.

Downloading Firmware

1. The MultiVoip Configuration program must be off when invoking the **Download Firmware** command. If it is on, the command will not work.
2. To invoke the Download Factory Defaults command, go to **Start** | **Programs** | **MVP____ x.xx** | **Download Firmware**.

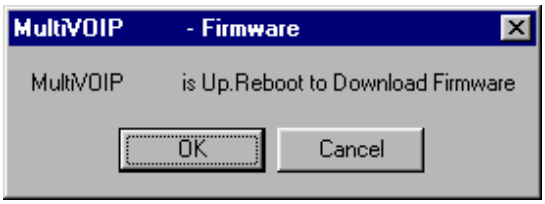


3. If a password has been established, the **Password Verification** screen will appear.



Type in the username and password and click **OK**.

4. The **MultiVOIP ___- Firmware** screen appears saying "MultiVOIP [*model number*] is up. Reboot to Download Firmware?"

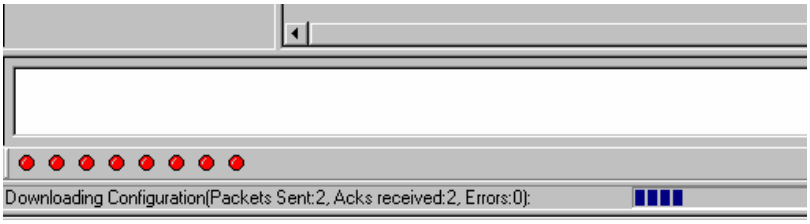


Click **OK** to download the firmware.

The "Boot" LED on the MultiVOIP will light up and remain lit during the file transfer process.

5. The program will locate the firmware ".bin" file in the MultiVOIP directory. Highlight the correct (newest) ".bin" file and click **Open**.

6. Progress bars will appear at the bottom of the screen during the file transfer.



The MultiVOIP's "Boot" LED will turn off at the end of the transfer.

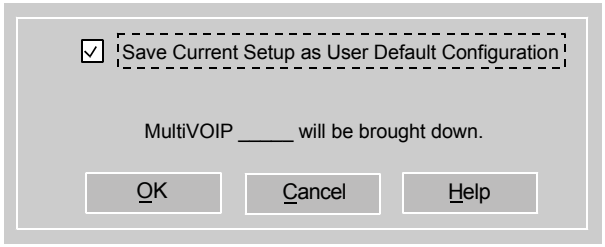
7. The **Download Firmware** procedure is complete.

Setting and Downloading User Defaults

The **Download User Defaults** command allows you to maintain a known working configuration that is specific to your VOIP system. You can then experiment with alterations or improvements to the configurations confident that a working configuration can be restored if necessary.

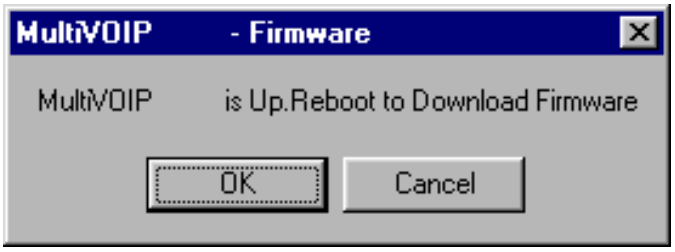
1. Before you can invoke the Download User Defaults command, you must first save a set of configuration parameters by using the **Save Setup** command in the sidebar menu of the MultiVOIP software.

2. Before the setup configuration is saved, you will be prompted to save the setup as the User Default Configuration. Select the checkbox and click **OK**.



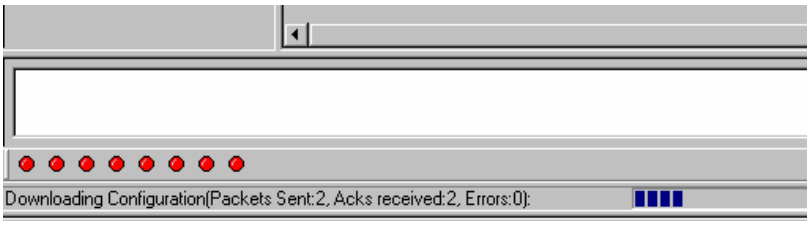
A user default file will be created.

3. The **MVP____- Firmware** screen appears saying "MultiVOIP [*model number*] is up. Reboot to Download Firmware?"

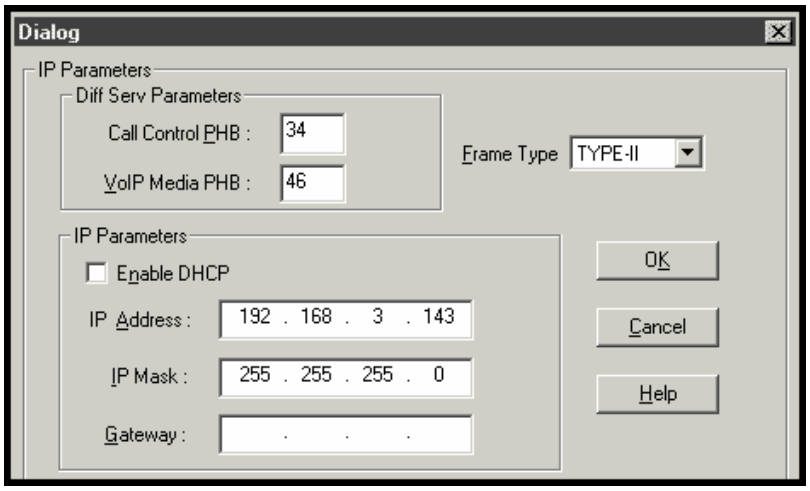


Click **OK** to download the factory defaults. The "Boot" LED on the MultiVOIP will light up and remain lit during the file transfer process.

4. Progress bars will appear at the bottom of the screen during the file transfer.



5. When the file transfer process is complete, the **Dialog-- IP Parameters** screen will appear.



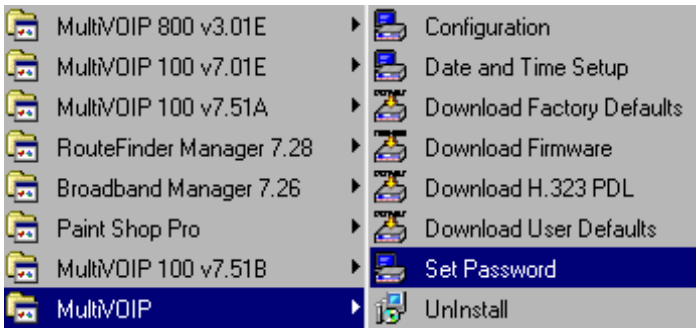
6. Set the IP values per your particular VOIP system. Click **OK**. Progress bars will appear as the MultiVOIP reboots itself.

Setting a Password (Windows GUI)

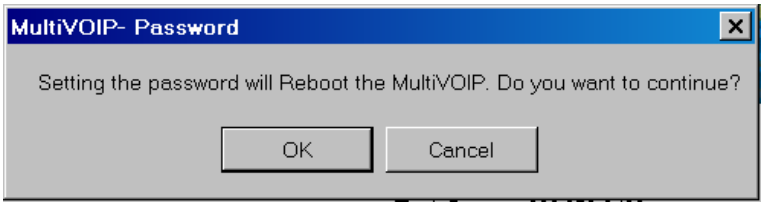
After a user name has been designated and a password has been set, that password is required to gain access to any functionality of the MultiVOIP software. Only one user name and password can be assigned to a voip unit. While a username and password are not absolutely required when communicating with the MultiVOIP via the web browser GUI, we highly recommend establishing username/password for security reasons.

NOTE: Record your user name and password in a safe place. If the password is lost, forgotten, or unretrievable, the user must contact MultiTech Tech Support in order to resume use of the MultiVOIP unit.

1. The MultiVoip configuration program must be off when invoking the **Set Password** command. If it is on, the command will not work.
2. To invoke the **Set Password** command, go to **Start** | **Programs** | **MVP____ x.xx** | **Set Password**.



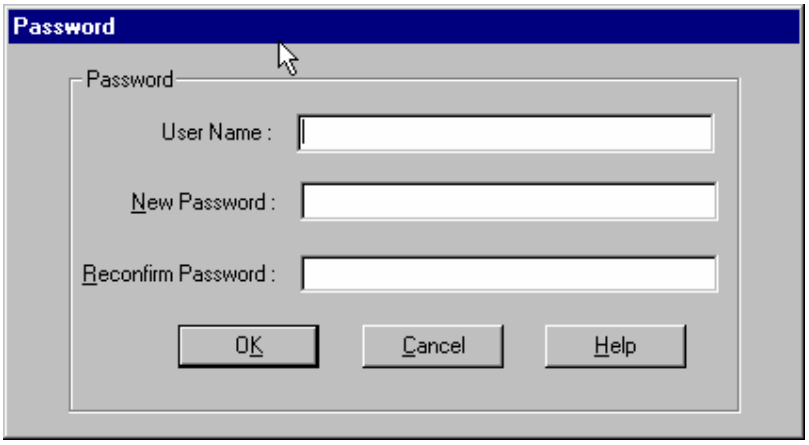
3. You will be prompted to confirm that you want to establish a password, which will entail rebooting the MultiVOIP (which is done automatically).



Click **OK** to proceed with establishing a password.

4. The **Password** screen will appear. If you intend to use the FTP Server function that is built into the MultiVOIP, enter a username and password. Type your password in the **Password** field of the **Password** screen. Type this same password again in the **Confirm Password** field to verify the password you have chosen.

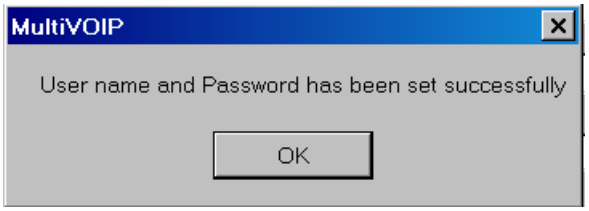
NOTE: Be sure to write down your password in a convenient but secure place. If the password is forgotten, contact MultiTech Technical Support for advice.



NOTE: You can disable the username and password functions by entering a username but then leaving both the "New Password" field and the "Reconfirm Password" field blank . Then click **OK**. In that case, you will not be prompted for username or password when entering the Windows GUI.

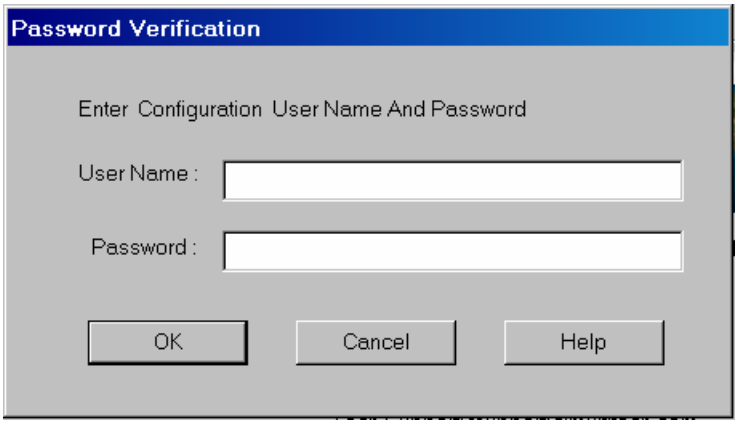
Click **OK**.

5. A message will appear indicating that a password has been set successfully.



After the password has been set successfully, the MultiVOIP will re-boot itself and, in so doing, its **BOOT** LED will light up.

6. After the password has been set, the user will be required to enter the User Name and Password to gain access to the web browser GUI and any part of the MultiVOIP software listed in the **Program** group menu. User Name and Password are both needed for access to the FTP Server residing in the MultiVOIP.



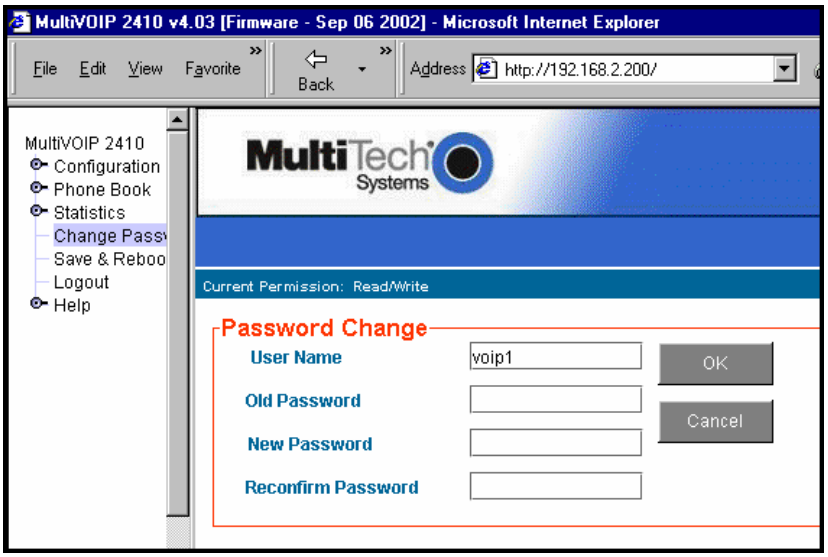
When MultiVOIP program asks for password at launch of program, the program will simply shut down if **CANCEL** is selected.

The MultiVOIP program will produce an error message if an invalid password is entered.

Setting a Password (Web Browser GUI)

Setting a password is optional when using the MultiVOIP web browser GUI. Only one password can be assigned and it works for all MultiVOIP software functions (Windows GUI, web browser GUI, FTP server, and all Program menu commands, e.g., Upgrade Software – only the FTP Server function requires a User Name in addition to the password). After a password has been set, that password is required to access the MultiVOIP web browser GUI.

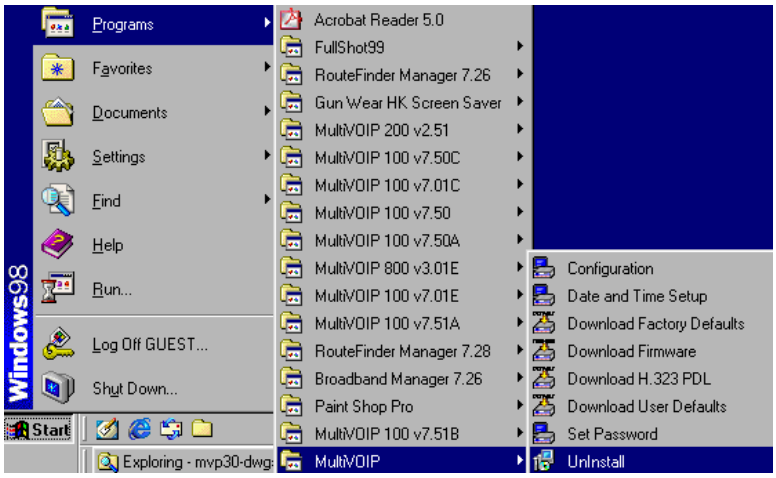
NOTE: Record your user name and password in a safe place. If the password is lost, forgotten, or unretrievable, the user must contact MultiTech Tech Support in order to resume use of the MultiVOIP web browser GUI.



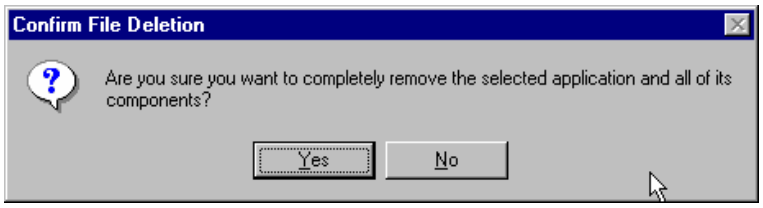
NOTE: You can disable the username and password functions by entering a username but then leaving both the "New Password" field and the "Reconfirm Password" field blank . Then click **OK**. In that case, you will not be prompted for username or password when entering the web browser GUI.

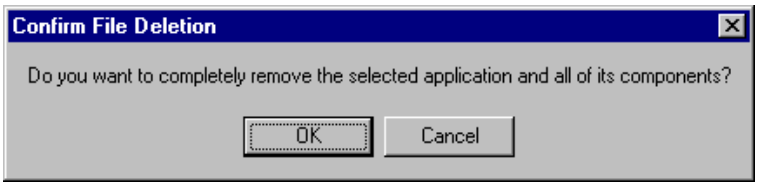
Un-Installing the MultiVOIP Software

1. To un-install the MultiVOIP configuration software, go to **Start** | **Programs** and locate the MultiVOIP entry. Select **Uninstall MVP____ vx.xx** (versions may vary).

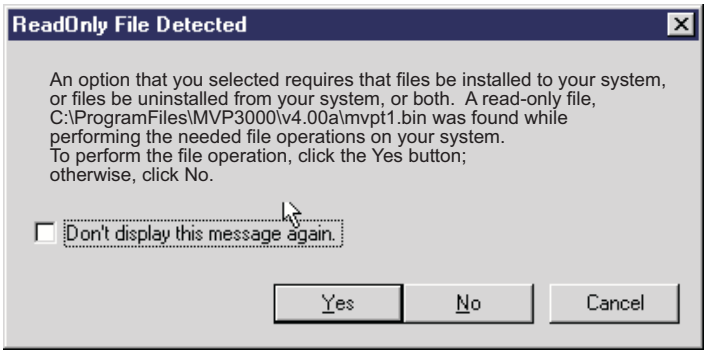


2. Two confirmation screens will appear. Click **Yes** and **OK** when you are certain you want to continue with the uninstallation process.

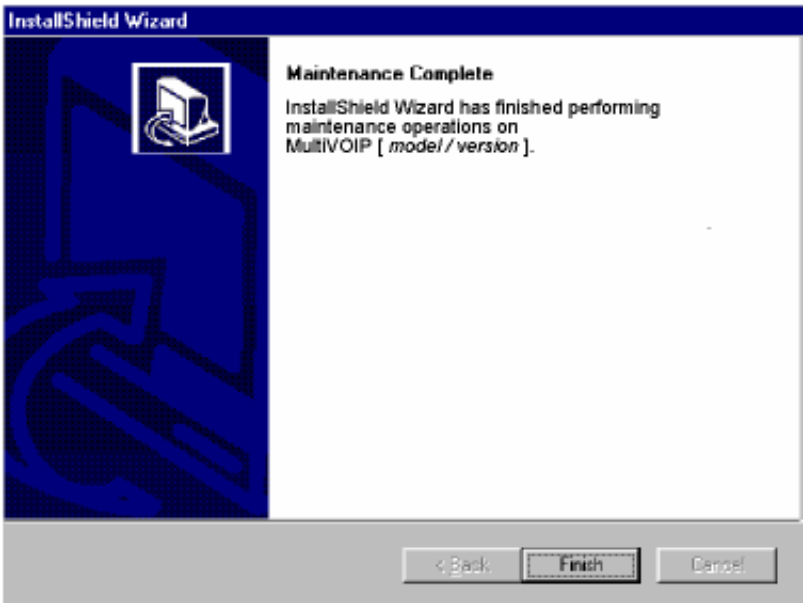

Confirm File Deletion
Are you sure you want to completely remove the selected application and all of its components?
Yes
No



3. A special warning message similar to that shown below may appear for the MultiVOIP software's ".bin" file. Click **Yes**.


ReadOnly File Detected
An option that you selected requires that files be installed to your system, or files be uninstalled from your system, or both. A read-only file, C:\ProgramFiles\MVP3000\v4.00a\mvpt1.bin was found while performing the needed file operations on your system.
To perform the file operation, click the Yes button; otherwise, click No.
Don't display this message again.
Yes
No
Cancel

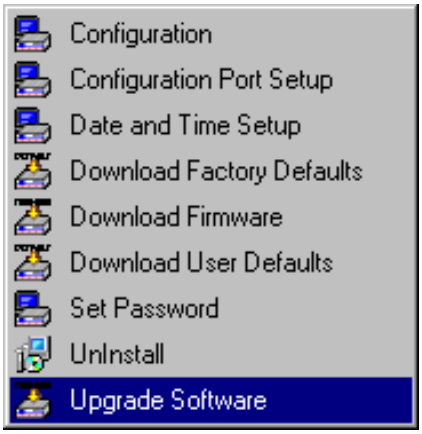
4. A completion screen will appear.



Click **Finish**.

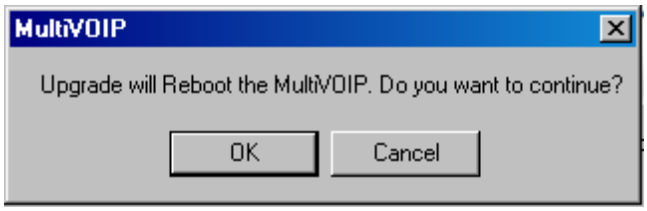
Upgrading Software

As noted earlier (see the section *Implementing a Software Upgrade* above), the Upgrade Software command transfers, from the controller PC to the MultiVOIP unit, firmware (including the H.323 stack) and factory default configuration settings. As such, **Upgrade Software** both implements the **Download Firmware** command and lets the user choose between retaining the current configuration values or downloading factory default values. 1. From the **Start** menu, invoke the **Upgrade Software** command.



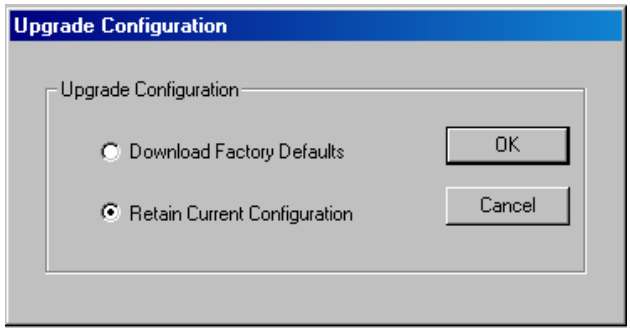
The Boot LED on the front panel of the MultiVOIP will come on.

2. A confirmation screen appears and indicates that upgrading the software will involve rebooting the voip.



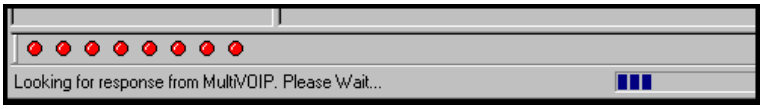
Click **OK** to proceed.

3. The **Upgrade Configuration** screen will appear . It allows you to choose which configuration settings you will apply to the upgraded software: your *current* configuration settings, or the factory default configuration settings.



Choose one and click **OK**.

4. You can monitor the progress of the upgrade process in the lower left corner of the screen.



5. When the booting process is done, the GUI will shut down.

FTP Server File Transfers ("Downloads")

Special FTP Upgrade Instructions for SW version 5.06 and earlier

To upgrade a MultiVOIP from software version 5.06 or earlier, an ftp primer file must first be sent to the VOIP. This file is located in the Software/ftp_Primer folder on the CD and the file name is "FTP_Primer.bin".

Before uploading this file, it must be renamed "mvpt1ftp.bin". The VoIP will only accept files of this name. This is a safety precaution to prevent the wrong files from being uploaded to the VoIP. Once the primer file has been uploaded, upload the FTP firmware file. If you accepted the defaults during the software loading process, this file is located on your local drive at C:\Program Files\Multi-Tech Systems\MultiVOIP 5.09. The firmware file itself is named 'mvpt1ftp.bin'.

Note 1: You cannot upgrade 5.04 or earlier versions using FTP. You must use 'upgradesoftware' via the serial port using the Windows GUI.

Note 2: These ftp upgrade instructions do not apply to software release 5.07 and above.

Note 3: These instructions apply equally whether you use an FTP client to upload files or use a web browser in conjunction with Windows Explorer to upload files.

General FTP File Transfer Information

MultiTech has built an FTP server into the MultiVOIP unit. Therefore, file transfers from the controller PC to the voip unit can be done using an FTP client program or even using a browser (e.g., Internet Explorer or Netscape, used in conjunction with Windows Explorer).

The terminology of "downloads" and "uploads" gets a bit confusing in this context. File transfers from a client to a server are typically considered "uploads." File transfers from a large repository of data to machines with less data capacity are considered "downloads." In this case, these metaphors are contradictory: the FTP server is actually housed in the MultiVOIP unit, and the controller PC, which is actually the repository of the info to be transferred, uses an FTP client program. In this situation, we have chosen to call the transfer of files from the PC to the voip "downloads." (Be aware that some FTP client programs may use the opposite terminology, i.e., they may refer to the file transfer as an "upload ")

You can download firmware, default configuration parameters, and phonebook data for the MultiVOIP unit with this FTP functionality. These downloads are done over a network, not by a local serial port connection. Consequently, voips at distant locations can be updated from a central control point.

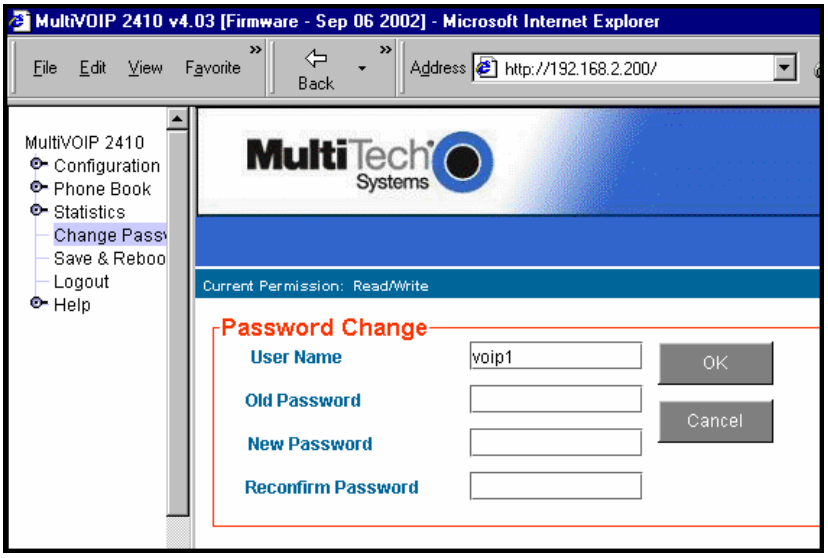
The phonebook downloading feature greatly reduces the data-entry required to establish inbound and outbound phonebooks for the voip units within a system. Although each MultiVOIP unit will require some unique phonebook entries, most will be common to the entire voip system. After the phonebooks for the first few voip units have been compiled, phonebooks for additional voips become much simpler: you copy the common material by downloading and then do data entry for the few phonebook items that are unique to that particular voip unit or voip site.

To transfer files using the FTP server functionality in the MultiVOIP, follow these directions.

1. **Establish Network Connection and IP Addresses**. Both the controller PC and the MultiVOIP unit(s) must be connected to the same IP network. An IP address must be assigned for each.

IP Address of Control PC	____ .	____ .	____ .	____
IP Address of voip unit #1	____ .	____ .	____ .	____
:	:	:	:	:
IP address of voip unit #n	____ .	____ .	____ .	____

2. **Establish User Name and Password**. You must establish a user name and (optionally) a password for contacting the voip over the IP network. (When connection is made via a local serial connection between the PC and the voip unit, no user name is needed.)

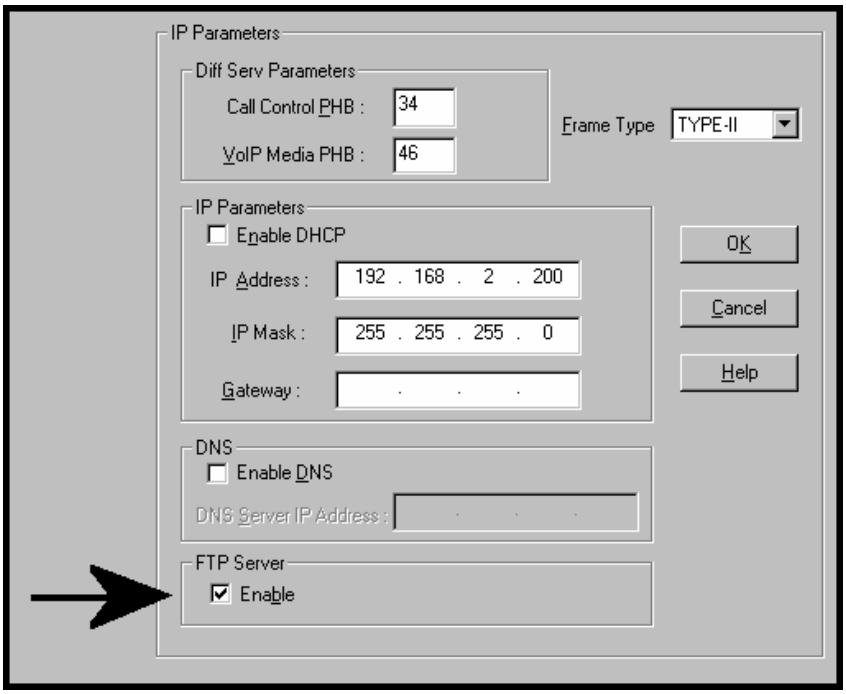


As shown above, the username and password can be set in the web GUI as well as in the Windows GUI.

3. **Install FTP Client Program or Use Substitute**. You *should* install an FTP client program on the controller PC. FTP file transfers can be done using a web browser (e.g., Netscape or Internet Explorer) in conjunction with a local Windows browser a (e.g., Windows Explorer), but this approach is somewhat clumsy (it requires use of two application programs rather than one) and it limits downloading to only one VOIP unit at a time. With an FTP client program, multiple voips can receive FTP file transmissions in response to a single command (the transfers may occur serially however).

Although MultiTech does not provide an FTP client program with the MultiVOIP software or endorse any particular FTP client program, we remind our readers that adequate FTP programs are readily available under retail, shareware and freeware licenses. (Read and observe any End-User License Agreement carefully.) Two examples of this are the "WSFTP" client and the "SmartFTP" client, with the former having an essentially text-based interface and the latter having a more graphically oriented interface, as of this writing. User preferences will vary. Examples here show use of both programs.

4. **Enable FTP Functionality**. Go to the **IP Parameters** screen and click on the "FTP Server: Enable" box.

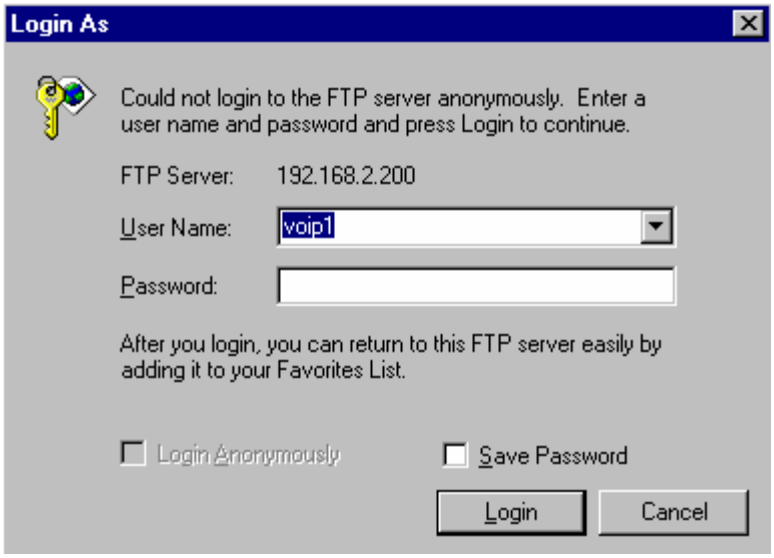


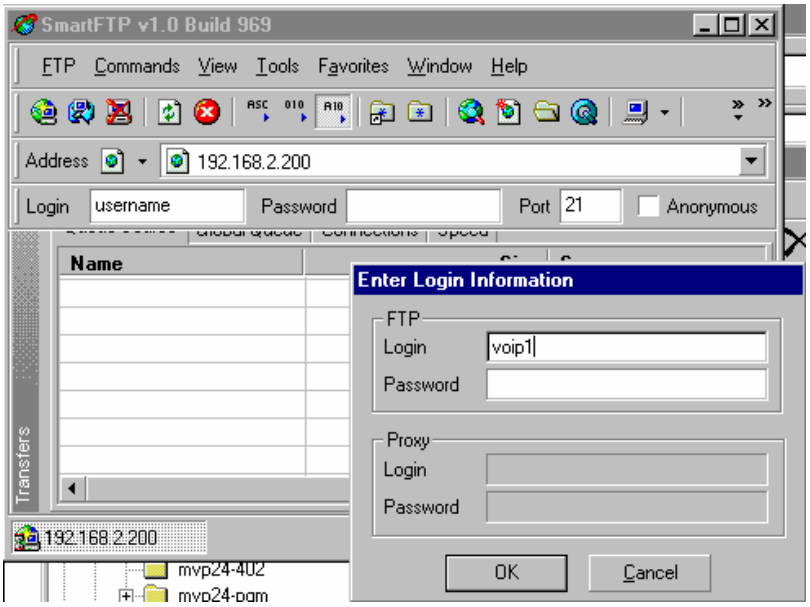
5. **Identify Files to be Updated**. Determine which files you want to update. Six types of files can be updated using the FTP feature. In some cases, the file to be transferred will have "Ftp" as the part of its filename just before the suffix (or extension). So, for example, the file "mvpt1Ftp.bin" can be transferred to update the bin file (firmware) residing in the MultiVOIP. Similarly, the file "fxo_loopFtp.cas" could be transferred to enable use of the FXO Loop Start telephony interface in one of the analog voip units and the file "r2_brazilFtp.cas" could be transferred to enable a particular telephony protocol used in Brazil.

File Type	File Names	Description
firmware "bin" file	mvpt1Ftp.bin	This is the MultiVOIP firmware file. Only one file of this type will be in the directory.
factory defaults	fdefFtp.cnf	This file contains factory default settings for user-changeable configuration parameters. Only one file of this type will be in the directory.
H323 PDL file		This file is specific to the particular version of the H.323 standard being used. This file rarely needs to be updated.
inbound phonebook	InPhBk.tmr	This file updates the inbound phonebook in the MultiVOIP unit.
outbound phonebook	OutPhBk.tmr	This file updates the outbound phonebook in the MultiVOIP unit.

6. **Contact MultiVOIP FTP Server**. You must make contact with the FTP Server in the voip using either a web browser or FTP client program. Enter the IP address of the MultiVOIP's FTP Server. If you are using a browser, the address must be preceded by "ftp://" (otherwise you'll reach the web GUI within the MultiVOIP unit).

7. **Log In**. Use the User Name and password established in item #2 above. The login screens will differ depending on whether the FTP file transfer is to be done with a web browser (see first screen below) or with an FTP client program (see second screen below).


Login As
Could not login to the FTP server anonymously. Enter a user name and password and press Login to continue.
FTP Server: 192.168.2.200
User Name: voip1
Password:
After you login, you can return to this FTP server easily by adding it to your Favorites List.
Login Anonymously
Save Password
Login
Cancel

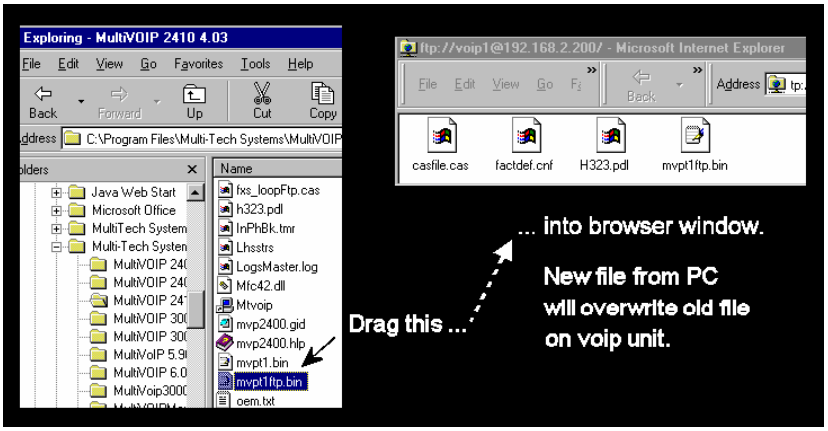


8. **Invoke Download**. Downloading can be done with a web browser or with an FTP client program.

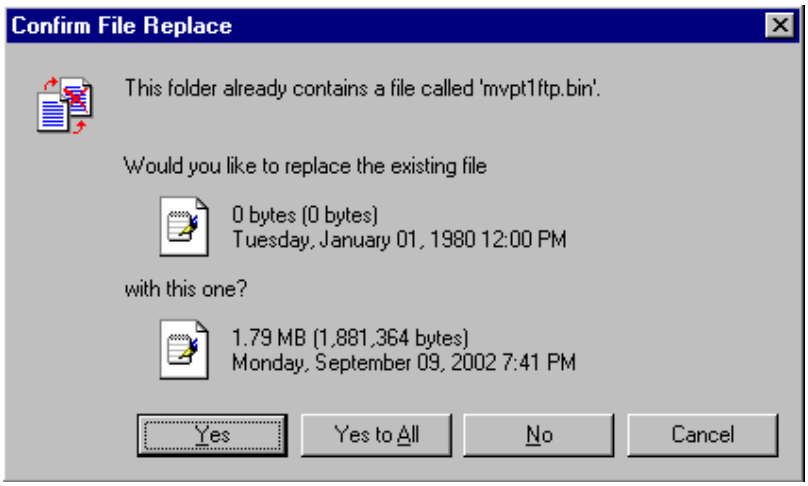
8A. Download with Web Browser.

8A1. In the local Windows browser, locate the directory holding the MultiVOIP program files. The default location will be C:\Program Files \Multi-Tech Systems \MultiVOIP xxxx yyyy (where x and y represent MultiVOIP model numbers and software version numbers).

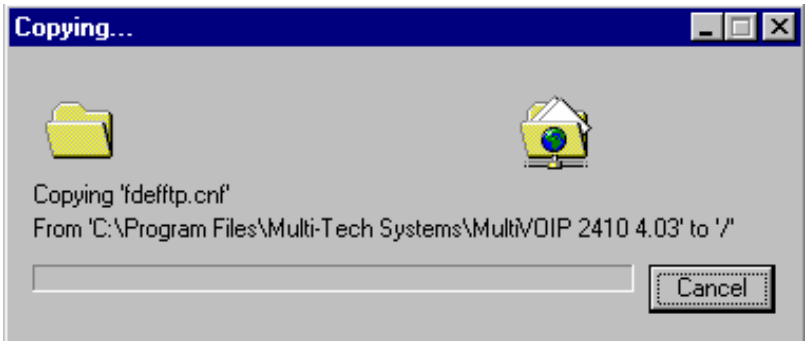
8A2. Drag-and-drop files from the local Windows browser (e.g., Windows Explorer) to the web browser.



You may be asked to confirm the overwriting of files on the MultiVOIP. Do so.


Confirm File Replace
This folder already contains a file called 'mvpt1ftp.bin'.
Would you like to replace the existing file
0 bytes (0 bytes)
Tuesday, January 01, 1980 12:00 PM
with this one?
1.79 MB (1,881,364 bytes)
Monday, September 09, 2002 7:41 PM
Yes
Yes to All
No
Cancel

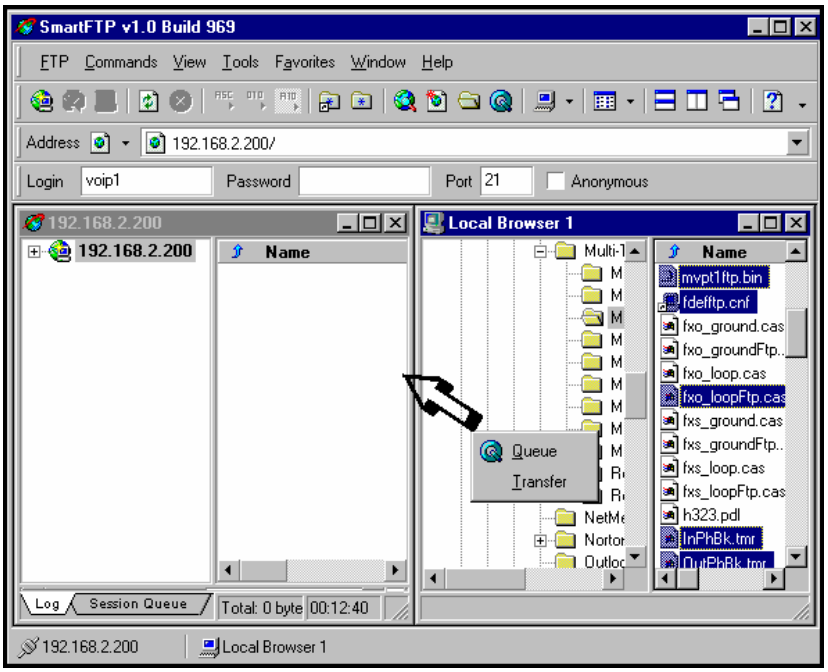
File transfer between PC and voip will look like transfer within voip directories.



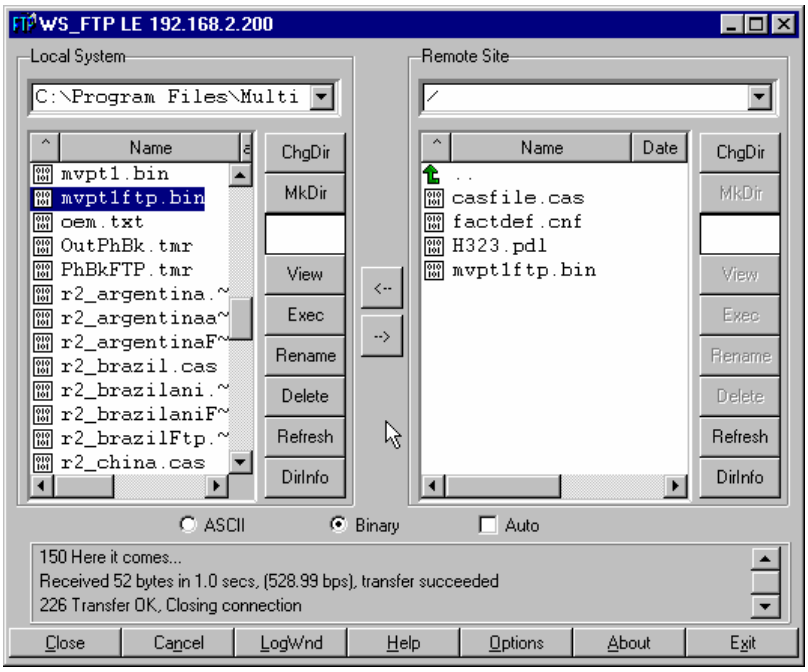
8B. Download with FTP Client Program.

8B1. In the local directory browser of the FTP client program, locate the directory holding the MultiVOIP program files. The default location will be C:\Program Files \Multi-Tech Systems \MultiVOIP xxxx yyyy (where x and y represent MultiVOIP model numbers and software version numbers).

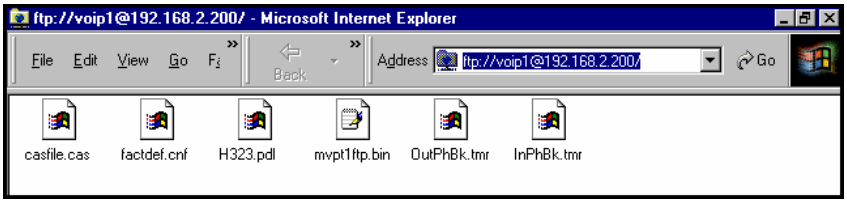
8B2. In the FTP client program window, drag-and-drop files from the local browser pane to the pane for the MultiVOIP FTP server. FTP client GUI operations vary. In some cases, you can choose between immediate and queued transfer. In some cases, there may be automated capabilities to transfer to multiple destinations with a single command.



Some FTP client programs are more graphically oriented (see previous screen), while others (like the "WS-FTP" client) are more text oriented.

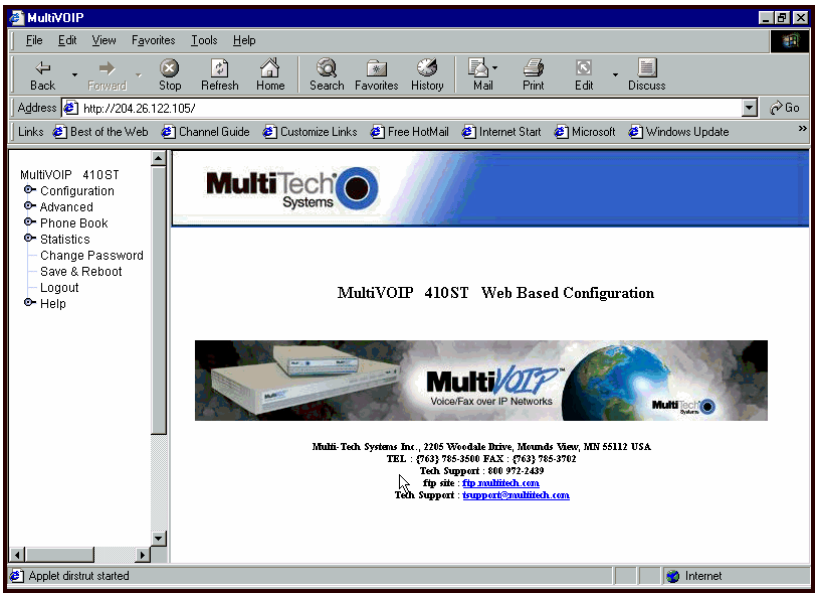


9. **Verify Transfer**. The files transferred will appear in the directory of the MultiVOIP.



10. **Log Out of FTP Session**. Whether the file transfer was done with a web browser or with an FTP client program, you *must* log out of the FTP session before opening the MultiVOIP Windows GUI.

Web Browser Interface

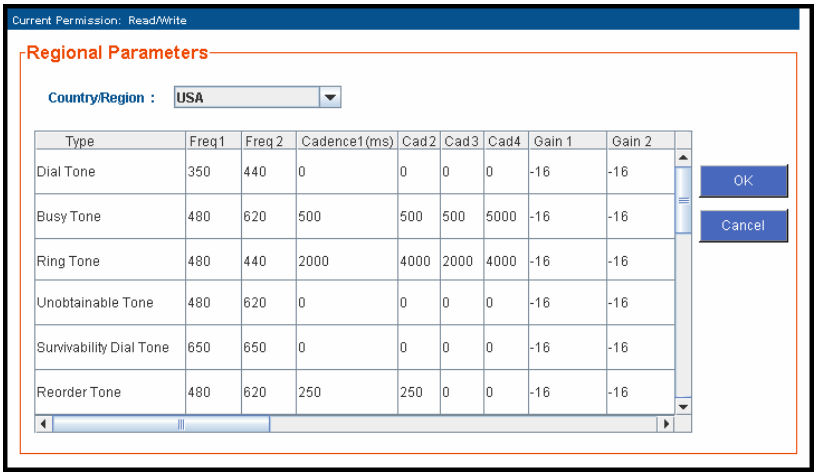


You can control the MultiVOIP unit with a graphic user interface (GUI) based on the common web browser platform. Qualifying browsers are InternetExplorer6 and Netscape6.

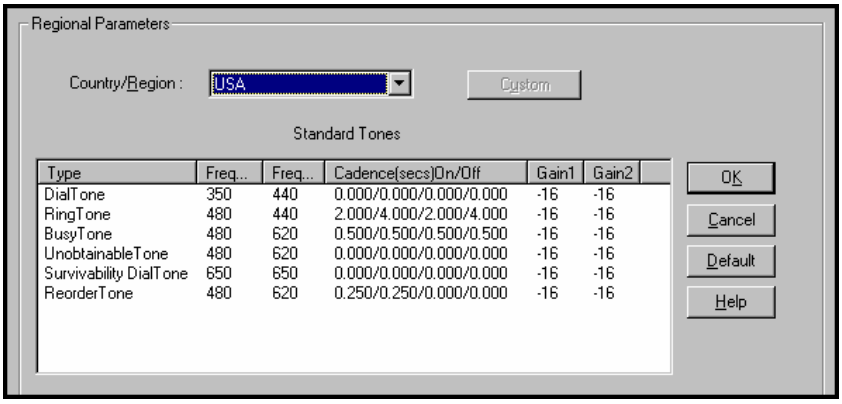
MultiVOIP Web Browser GUI Overview	
Function	Remote configuration and control of MultiVOIP units.
Configuration Prerequisite	Local Windows GUI must be used to assign IP address to MultiVOIP.
Browser Version Requirement	Internet Explorer 6.0 or higher; or Netscape 6.0 or higher
Java Requirement	Java Runtime Environment version 1.4.0_01 or higher (this application program is included with MultiVOIP)
Video Usability	large video monitor recommended

The initial configuration step of assigning the voip unit an IP address must still be done locally using the Windows GUI. However, all additional configuration can be done via the web GUI.

The content and organization of the web GUI is directly parallel to the Windows GUI. For each screen in the Windows GUI, there is a corresponding screen in the web GUI. The fields on each screen are the same, as well.


Current Permission: Read/Write
Regional Parameters
Country/Region : USA
Type Freq 1 Freq 2 Cadence1 (ms) Cad 2 Cad 3 Cad 4 Gain 1 Gain 2
Dial Tone 350 440 0 0 0 0 -16 -16
Busy Tone 480 620 500 500 500 5000 -16 -16
Ring Tone 480 440 2000 4000 2000 4000 -16 -16
Unobtainable Tone 480 620 0 0 0 0 -16 -16
Survivability Dial Tone 650 650 0 0 0 0 -16 -16
Reorder Tone 480 620 250 250 0 0 -16 -16
OK
Cancel

The Windows GUI gives access to commands via pulldown menus, sidebar menus, keyboard shortcuts, and, in some cases via command icons, whereas the web GUI has sidebar menus only.



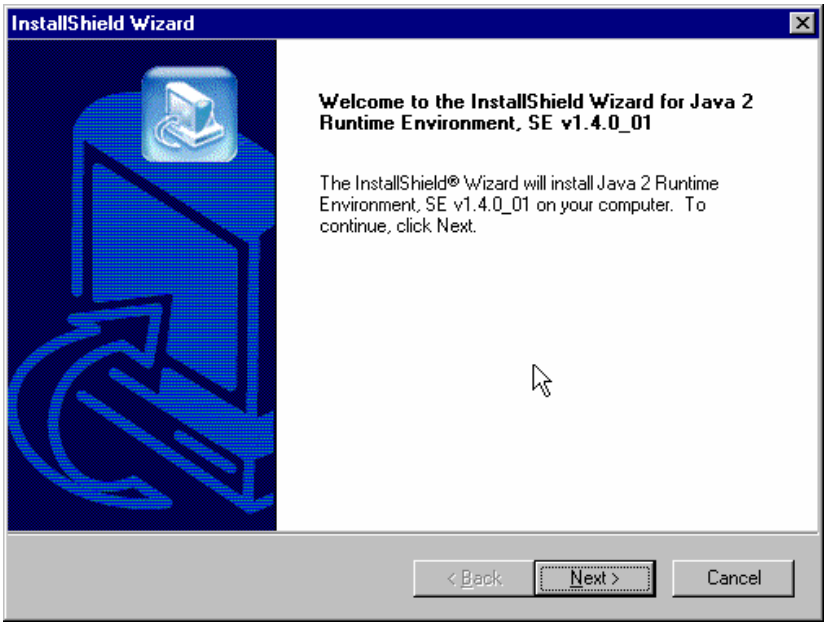
The web GUI, however, cannot perform logging in the same direct mode done in the Windows GUI. However, when the web GUI is used, logging can be done by email (SMTP).

The graphic layout of the web GUI is also somewhat larger-scale than that of the Windows GUI. For that reason, it's helpful to use as large of a video monitor as possible.

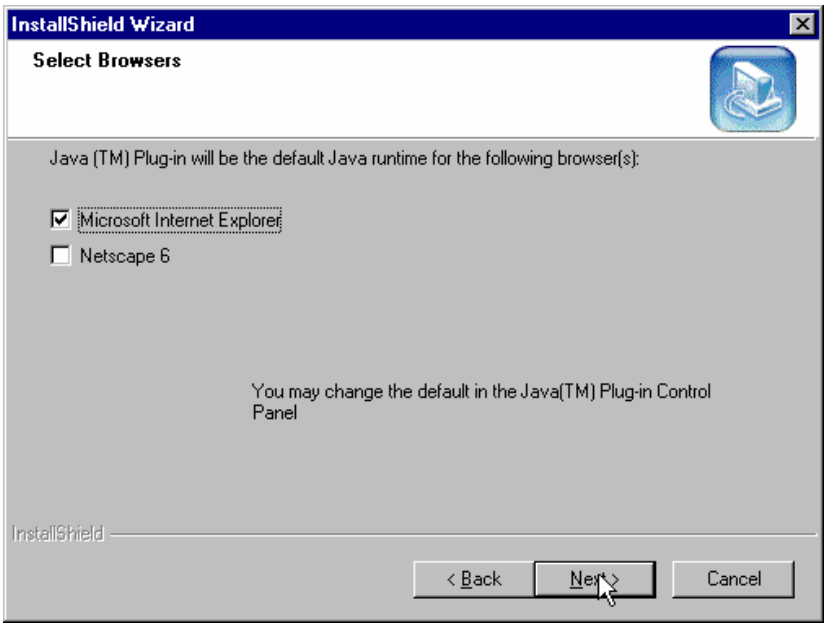
The primary advantage of the web GUI is remote access for control and configuration. The controller PC and the MultiVOIP unit itself must both be connected to the same IP network and their IP addresses must be known.

In order to use the web GUI, you must also install a Java application program on the controller PC. This Java program is included on the MultiVOIP product CD.). Java is needed to support drop-down menus and multiple windows in the web GUI.

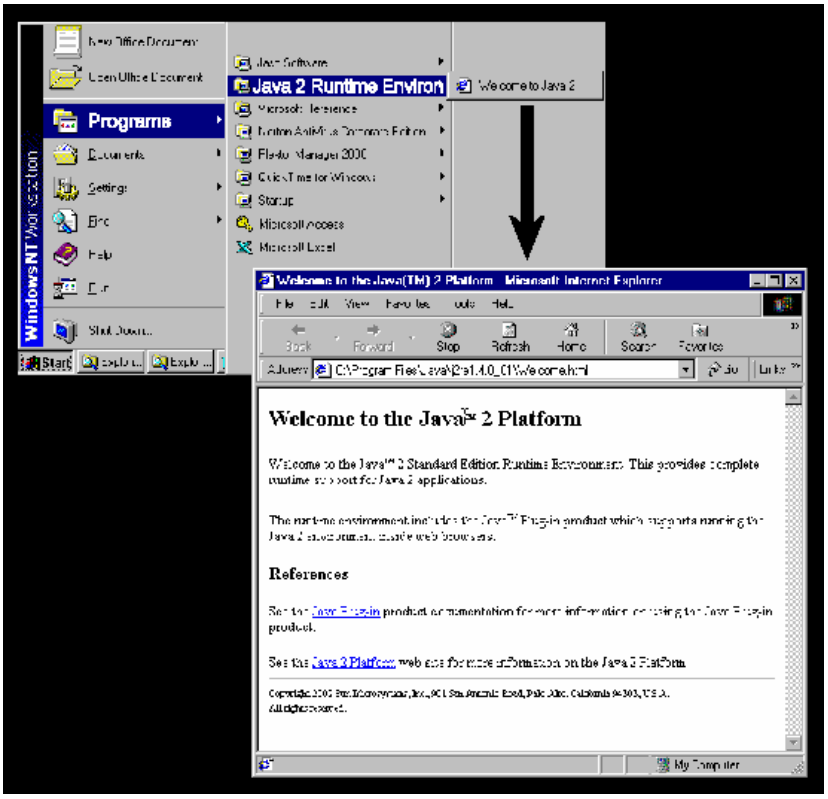
To install the Java program, go to the **Java** directory on the MultiVOIP product CD. Double-click on the EXE file to begin the installation. Follow the instructions on the Install Shield screens.


InstallShield Wizard
Welcome to the InstallShield Wizard for Java 2 Runtime Environment, SE v1.4.0_01
The InstallShield® Wizard will install Java 2 Runtime Environment, SE v1.4.0_01 on your computer. To continue, click Next.
< Back
Next >
Cancel

During the installation, you must specify which browser you'll use in the **Select Browsers** screen.


InstallShield Wizard
Select Browsers
Java (TM) Plug-in will be the default Java runtime for the following browser(s):
Microsoft Internet Explorer
Netscape 6
You may change the default in the Java(TM) Plug-in Control Panel
InstallShield
< Back
Next >
Cancel

When installation is complete, the Java program becomes accessible in your **Start | Programs** menu (Java resources are readily available via the web). However, the Java program runs automatically in the background as a plug-in supporting the MultiVOIP web GUI. No overt user actions are required.


Java 2 Runtime Environ
Welcome to Java 2
Programs
Welcome to the Java(TM) 2 Platform - Microsoft Internet Explorer
Welcome to the Java™ 2 Platform
References

After the Java program has been installed, you can access the MultiVOIP using the web browser GUI. Close the MultiVOIP Windows GUI. Start the web browser. Enter the IP address of the MultiVOIP unit. Enter a password when prompted. (A password is needed here only if password has been set for the local Windows GUI or for the MultiVOIP's FTP Server function. See "Setting a Password -- Web Browser GUI" earlier in this chapter.) The web browser GUI offers essentially the same control over the voip as can be achieved using the Windows GUI. As noted earlier, logging functions cannot be handled via the web GUI. And, because network communications will be slower than direct communications over a serial PC cable, command execution will be somewhat slower over the web browser GUI than with the Windows GUI.

SysLog Server Functions

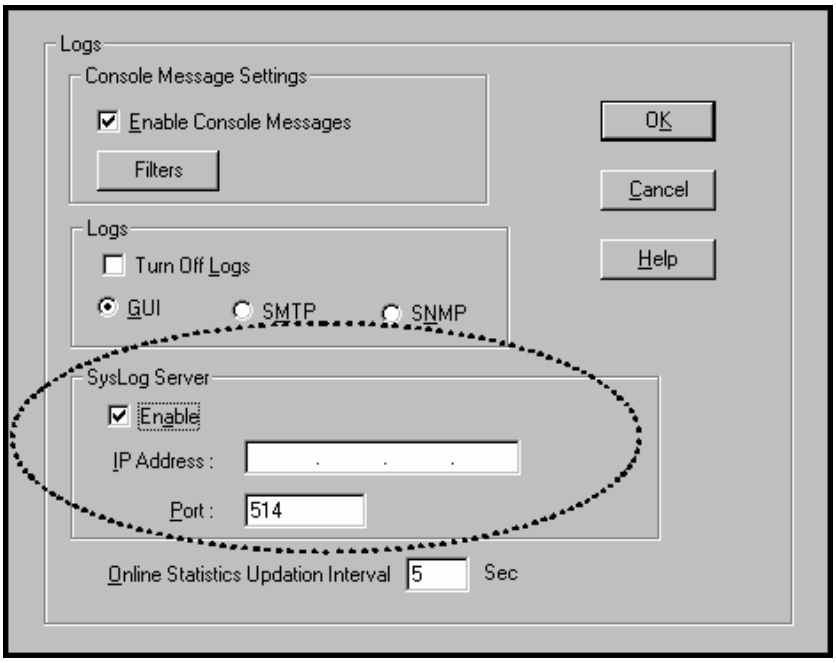
MultiTech has built SysLog server functionality into the software of the MultiVOIP units. SysLog is a *de facto* standard for logging events in network communication systems.

The SysLog Server resides in the MultiVOIP unit itself. To implement this functionality, you will need a SysLog client program (sometimes referred to as a "daemon"). SysLog client programs, both paid and freeware, can be obtained from Kiwi Enterprises, among other firms. Read the End-User License Agreement carefully and observe license requirements. See www.kiwisyslog.com. SysLog client programs essentially give you a means of structuring console messages for convenience and ease of use.

MultiTech Systems does not endorse any particular SysLog client program. SysLog client programs by qualified providers should suffice for use with MultiVOIP units. Kiwi's brief description of their SysLog program is as follows:

> "Kiwi Syslog Daemon is a freeware Syslog Daemon for the Windows platform. It receives, logs, displays and forwards Syslog messages from hosts such as routers, switches, Unix hosts and any other syslog enabled device. There are many customizable options available."

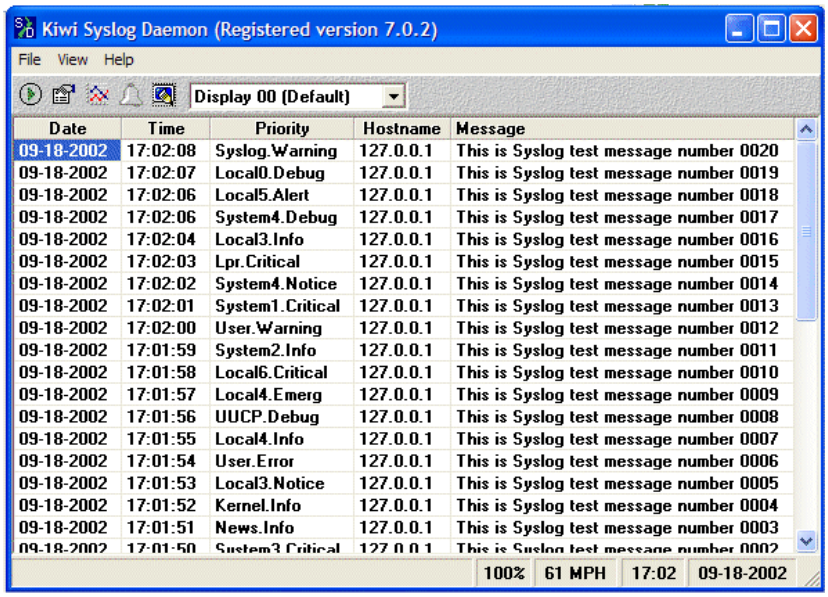
Before a SysLog client program is used, the SysLog functionality must be enabled within the MultiVOIP in the **Logs** menu under **Configuration**.



The IP Address used will be that of the MultiVOIP itself.

In the **Port** field, entered by default, is the standard ('well-known') logical port, 514.

Configuring the SysLog Client Program. Configure the SysLog client program for your own needs. In various SysLog client programs, you can define where log messages will be saved/archived, opt for interaction with an SNMP system (like MultiVoipManager), set the content and format of log messages, determine disk space allocation limits for log messages, and establish a hierarchy for the seriousness of messages (normal, alert, critical, emergency, etc.). A sample presentation of SysLog info in the Kiwi daemon is shown below. SysLog programs will vary in features and presentation.


Kiwi Syslog Daemon (Registered version 7.0.2)
File View Help
Display 00 (Default)
Date Time Priority Hostname Message
09-18-2002 17:02:08 Syslog.Warning 127.0.0.1 This is Syslog test message number 0020
09-18-2002 17:02:07 Local0.Debug 127.0.0.1 This is Syslog test message number 0019
09-18-2002 17:02:06 Local5.Alert 127.0.0.1 This is Syslog test message number 0018
09-18-2002 17:02:06 System4.Debug 127.0.0.1 This is Syslog test message number 0017
09-18-2002 17:02:04 Local3.Info 127.0.0.1 This is Syslog test message number 0016
09-18-2002 17:02:03 Lpr.Critical 127.0.0.1 This is Syslog test message number 0015
09-18-2002 17:02:02 System4.Notice 127.0.0.1 This is Syslog test message number 0014
09-18-2002 17:02:01 System1.Critical 127.0.0.1 This is Syslog test message number 0013
09-18-2002 17:02:00 User.Warning 127.0.0.1 This is Syslog test message number 0012
09-18-2002 17:01:59 System2.Info 127.0.0.1 This is Syslog test message number 0011
09-18-2002 17:01:58 Local6.Critical 127.0.0.1 This is Syslog test message number 0010
09-18-2002 17:01:57 Local4.Emerg 127.0.0.1 This is Syslog test message number 0009
09-18-2002 17:01:56 UUCP.Debug 127.0.0.1 This is Syslog test message number 0008
09-18-2002 17:01:55 Local4.Info 127.0.0.1 This is Syslog test message number 0007
09-18-2002 17:01:54 User.Error 127.0.0.1 This is Syslog test message number 0006
09-18-2002 17:01:53 Local3.Notice 127.0.0.1 This is Syslog test message number 0005
09-18-2002 17:01:52 Kernel.Info 127.0.0.1 This is Syslog test message number 0004
09-18-2002 17:01:51 News.Info 127.0.0.1 This is Syslog test message number 0003
100% 61 MPH 17:02 09-18-2002

Chapter 9 Warranty, Service, and Tech Support

Limited Warranty

Multi-Tech Systems, Inc. ("MTS") warrants that its products will be free from defects in material or workmanship for a period of two years from the date of purchase, or if proof of purchase is not provided, two years from date of shipment. MTS MAKES NO OTHER WARRANTY, EXPRESSED OR IMPLIED, AND ALL IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE HEREBY DISCLAIMED. This warranty does not apply to any products which have been damaged by lightning storms, water, or power surges or which have been neglected, altered, abused, used for a purpose other than the one for which they were manufactured, repaired by the customer or any party without MTS's written authorization, or used in any manner inconsistent with MTS's instructions.

MTS's entire obligation under this warranty shall be limited (at MTS's option) to repair or replacement of any products which prove to be defective within the warranty period, or, at MTS's option, issuance of a refund of the purchase price. Defective products must be returned by Customer to MTS's factory—transportation prepaid.

MTS WILL NOT BE LIABLE FOR CONSEQUENTIAL DAMAGES AND UNDER NO CIRCUMSTANCES WILL ITS LIABILITY EXCEED THE PURCHASE PRICE FOR DEFECTIVE PRODUCTS.

Repair Procedures for U.S. and Canadian Customers

In the event that service is required, products may be shipped, freight prepaid, to our Mounds View, Minnesota factory:

Multi-Tech Systems, Inc.
2205 Woodale Drive
Mounds View, MN 55112
Attn: Repairs, Serial # _____

A Returned Materials Authorization (RMA) is not required. Return shipping charges (surface) will be paid by MTS.

Please include, inside the shipping box, a description of the problem, a return shipping address (it must be a street address, not a P.O. Box number), your telephone number, and if the product is out of warranty, a check or purchase order for repair charges.

For out-of-warranty repair charges, go to www.multitech.com/documents/warranties

Extended two-year overnight replacement service agreements are available for selected products. Please call MTS at (888) 288-5470, extension 5308, or visit our web site at www.multitech.com/programs/orc for details on rates and coverages.

Please direct your questions regarding technical matters, product configuration, verification that the product is defective, etc., to our Technical Support department at (800) 972-2439 or email tsupport@multitech.com. Please direct your questions regarding repair expediting, receiving, shipping, billing, etc., to our Repair Accounting department at (800) 328-9717 or (763) 717-5631, or email mtsrepair@multitech.com.

Repairs for damages caused by lightning storms, water, power surges, incorrect installation, physical abuse, or used-caused damages are billed on a time-plus-materials basis.

Technical Support

Multi-Tech Systems has an excellent staff of technical support personnel available to help you get the most out of your Multi-Tech product. If you have any questions about the operation of this unit, or experience difficulty during installation you can contact Tech Support via the following:

Contacting Technical Support

Country	By E-mail	By telephone
France	support@multitech.fr	(33) 1-64 61 09 81
India	support@ multitechindia.com	(91) 124-340778
U.K.	support@ multitech.co.uk	(44) 118 959 7774
U.S. & Canada	tsupport@ multitech.com	(800) 972-2439
Rest of World	support@ multitech.com	(763) 785-3500

Internet: http://www.multitech.com/ **_forms/email_tech_support.htm**

Please have your product information available, including model and serial number.

Chapter 10: Regulatory Information

CE

EMC, Safety, and R&TTE Directive Compliance

The CE mark is affixed to this product to confirm compliance with the following European Community Directives:

Council Directive 89/336/EEC of 3 May 1989 on the approximation of the laws of Member States relating to electromagnetic compatibility,
and
Council Directive 73/23/EEC of 19 February 1973 on the harmonization of the laws of Member States relating to electrical equipment designed for use within certain voltage limits,
and
Council Directive 1999/5/EC of 9 March 1999 on radio equipment and telecommunications terminal equipment and the mutual recognition of their conformity.

FCC Declaration

NOTE: This equipment has been tested and found to comply with the limits for a **Class A** digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy, and if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

This device complies with Part 15 of the FCC rules.

Operation is subject to the following two conditions:

(1) This device may not cause harmful interference.

(2) This device must accept any interference that may cause undesired operation.

Warning: Changes or modifications to this unit not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

Industry Canada

This Class A digital apparatus meets all requirements of the Canadian Interference-Causing Equipment Regulations.

Cet appareil numérique de la classe A
respecte toutes les exigences du
Reglement Canadien sur le matériel brouilleur.

FCC Part 68 Telecom

1. This equipment complies with part 68 of the Federal Communications Commission Rules. On the outside surface of this equipment is a label that contains, among other information, the FCC registration number. This information must be provided to the telephone company.
2. As indicated below, the suitable jack (Universal Service Order Code connecting arrangement) for this equipment is shown. If applicable, the facility interface codes (FIC) and service order codes (SOC) are shown.
3. An FCC compliant telephone cord and modular plug is provided with this equipment. This equipment is designed to be connected to the telephone network or premises wiring using a compatible modular jack that is Part 68 compliant. See installation instructions for details.
4. If this equipment causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. If advance notice is not practical, the telephone company will notify the customer as soon as possible.
5. The telephone company may make changes in its facilities, equipment, operation, or procedures that could affect the operation of the equipment. If this happens, the telephone company will provide advance notice to allow you to make necessary modifications to maintain uninterrupted service.
6. If trouble is experienced with this equipment (the model of which is indicated below), please contact Multi-Tech Systems, Inc. at the address shown below for details of how to have repairs made. If the equipment is causing harm to the network, the telephone company

may request you to remove the equipment form t network until the problem is resolved.

7. No repairs are to be made by you. Repairs are to be made only by Multi-Tech Systems or its licensees. Unauthorized repairs void registration and warranty.

8. Manufacturer: Multi-Tech Systems, Inc.
   Trade name: MultiVOIP
   Model number: MVP-810ST/410ST
   FCC registration number: US: AU7DDNAN46050
   Modular jack (USOC): RJ-48C
   Service center in USA: Multi-Tech Systems, Inc.
   2205 Woodale Drive
   Mounds View, MN 55112
   Tel: (763) 785-3500
   FAX: (763) 785-9874

Canadian Limitations Notice

Notice: The Industry Canada label identifies certified equipment. This certification means that the equipment meets certain telecommunications network protective, operational and safety requirements. The Department does not guarantee the equipment will operate to the user's satisfaction.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs to certified equipment should be made by an authorized Canadian maintenance facility designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

Caution: Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.

WEEE Statement

(Waste Electrical and Electronic Equipment)

July, 2005

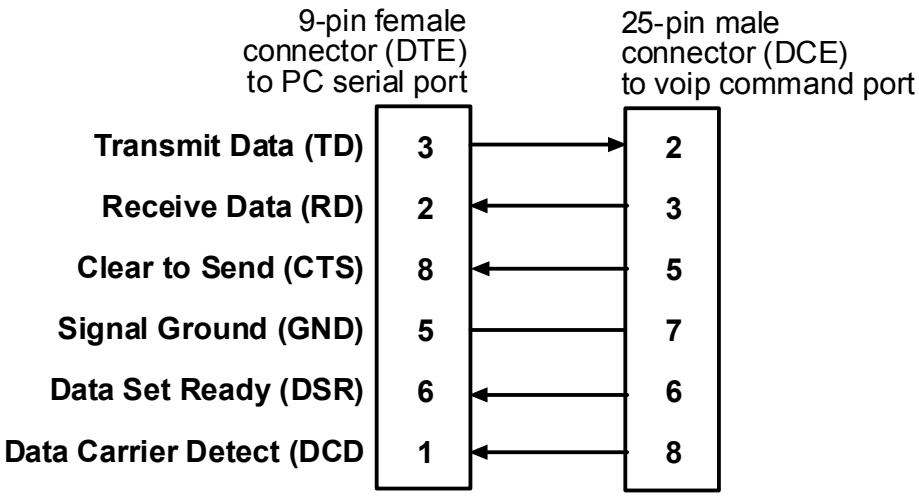
The WEEE directive places an obligation on EU-based manufacturers, distributors, retailers and importers to take-back electronics products at the end of their useful life. A sister Directive, ROHS (Restriction of Hazardous Substances) compliments the WEEE Directive by banning the presence of specific hazardous substances in the products at the design phase. The WEEE Directive covers all Multi-Tech products imported into the EU as of August 13, 2005. EU-based manufacturers, distributors, retailers and importers are obliged to finance the costs of recovery from municipal collection points, reuse, and recycling of specified percentages per the WEEE requirements.

Instructions for Disposal of WEEE by Users in the European Union
The symbol shown below is on the product or on its packaging, which indicates that this product must not be disposed of with other waste. Instead, it is the user's responsibility to dispose of their waste equipment by handing it over to a designated collection point for the recycling of waste electrical and electronic equipment. The separate collection and recycling of your waste equipment at the time of disposal will help to conserve natural resources and ensure that it is recycled in a manner that protects human health and the environment. For more information about where you can drop off your waste equipment for recycling, please contact your local city office, your household waste disposal service or where you purchased the product.

Appendix A: Cable Pinouts

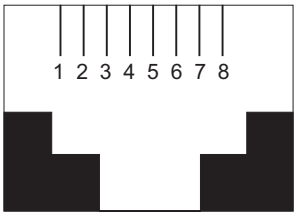
Appendix A: Cable Pinouts

Command Cable



Ethernet Connector

The functions of the individual conductors of the MultiVOIP's Ethernet port are shown on a pin-by-pin basis below.

RJ-45 Ethernet Connector	Pin	Circuit Signal Name
	1	TD+ Data Transmit Positive
	2	TD- Data Transmit Negative
	3	RD+ Data Receive Positive
	6	RD- Data Receive Negative

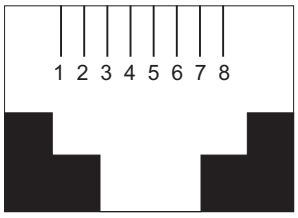
Pin Functions (FXS/FXO Interface)			
FXS Pin	**Description**	**FXO Pin**	**Description**
2	N/C	2	N/C
3	Ring	3	Tip
4	Tip	4	Ring
5	N/C	5	N/C

ISDN BRI RJ-45 Pinout Information

The S/T interface uses an 8-conductor modular cable terminated with an 8-pin RJ-45 plug. An 8-pin RJ-45 jack located on the terminal is used to connect the terminal to the DSL (Digital Subscriber Loops) using this modular cable.

The table below shows the Pin Number, Terminal Pin Signal Name and Network Pin Signal name for the S/T interface.

Pin	TE Signal		NT Signal	Pin
1	Not used		Not used	1
2	Not used		Not used	2
3	Tx+		Rx+	3
4	Rx-		Tx-	4
5	Rx+		Tx+	5
6	Tx-		Rx-	6
7	Not used		Not used	7
8	Not used		Not used	8



TE=Terminal Equipment

NT=Network

ISDN Interfaces: "ST" and "U"

The MVP410ST and MVP810ST are ISDN-BRI voip units that use an S/T outlet interface. You will need an NT1 device to connect these units to any network equipment that has the "U" ISDN interface. In the UK, and in many European countries, the telco supplies an NT1 device for ISDN-BRI service.

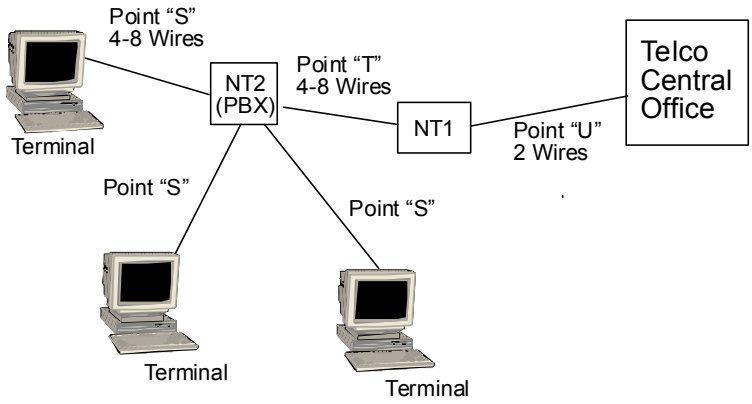
An ISDN Basic Rate (BRI) U-Loop consists of two conductors from the telco central office to the customer premises. The equipment on both sides of the U-loop accommodates the extensive length of the U-loop and the noisy environment in which it may operate. At the customer premises, the U-loop is terminated by an NT1 (network termination 1) device. An NT1 device makes an end-user's 4-wire terminal equipment compatible with the telco's 2-wire twisted pair ISDN-BRI line.

The NT1 drives an S/T bus. The S/T bus is usually made up of 4 wires, but in some cases may be 6 or 8 wires.

"S" and "T" refer to connection points in the ISDN specification.

When a PBX is present, *S* refers to the connection between the PBX and the terminal. ("Terminal" can mean any sort of end-user ISDN device: data terminals, telephones, FAX machines, voip units, etc.)

Point *T* refers to the connection between the NT1 device and customer supplied equipment. Terminals can connect directly to the NT1 device at point *T*, or there may be a PBX (private branch exchange, i.e., a customer-owned telephone exchange). The figure below shows "S" and "T" connection points in an ISDN network.



Appendix B: TCP/UDP Port Assignments

Well Known Port Numbers

The following description of port number assignments for Internet Protocol (IP) communication is taken from the Internet Assigned Numbers Authority (IANA) web site (www.iana.org).

> "The Well Known Ports are assigned by the IANA and on most systems can only be used by system (or root) processes or by programs executed by privileged users. Ports are used in the TCP [RFC793] to name the ends of logical connections which carry long term conversations. For the purpose of providing services to unknown callers, a service contact port is defined. This list specifies the port used by the server process as its contact port. The contact port is sometimes called the "well-known port". To the extent possible, these same port assignments are used with the UDP [RFC768]. The range for assigned ports managed by the IANA is 0-1023."

Well-known port numbers especially pertinent to MultiVOIP operation are listed below.

Port Number Assignment List

Well-Known Port Numbers

Function	Port Number
telnet	23
tftp	69
snmp	161
snmp tray	162
gatekeeper registration	1719
H.323	1720
SIP	5060
SysLog	514

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